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(54) Title: SECTORIZED COMMUNICATION SYSTEM AND METHODS USEFUL THEREFOR

(57) Abstract

Apparatus for providing voice and data communications in a frequency hopping multiple access communication system, preferably including automatic gain control apparatus, time alignment apparatus, power control apparatus, apparatus for coordinated hits and forced erasures, automatic frequency control apparatus, delay lock loop apparatus, apparatus for handling fringe areas, apparatus for channel feature acquisition and tracking, hand off apparatus, talk around apparatus, apparatus providing service at the foot of a base station, and apparatus for retransmission of messages.

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SECTORIZED COMMUNICATION SYSTEM AND METHODS USEFUL THEREFOR

FIELD OF THE INVENTION

The present invention relates to apparatus and method for providing voice and data communications. More specifically, it relates to apparatus and method for providing a frequency hopping multiple access communication system.

BACKGROUND OF THE INVENTION

Multiple access communications systems are capable of providing multiple communications at the same time using the same system resources. These systems utilize various communications protocol and various system architectures. One protocol is time division multiple access wherein users communicate on a shared channel at different times. Another protocol is frequency division multiple access wherein separate frequency channels are allocated to mobile radio terminals. The system architecture commonly utilized is a cellular type architecture wherein there are many base stations that provide communication channels to many radio terminals.

The existing multiple access communication systems all have various drawbacks. By way of example only, many rely on a hardware intensive architecture that is costly to implement and also costly to operate. These high costs result in a higher cost to customers. The existing multiple access communication systems also have severe limitations on system capacity or, stated differently, require a great deal of spectrum to efficiently operate. The existing multiple access communication systems also have limitations on the quality and the types of the communications services

provided, particularly when the system is fully loaded.

Thus, a communication system that offers low cost implementation and operation along with improved communication characteristics is needed.

Typical protocols used in multiple access communication systems include time division multiple access (TDMA), frequency division multiple access (FDMA), and frequency hopping multiple access (FHMA).

In systems employing TDMA protocols, users communicate on a shared channel but at different times. In systems employing FDMA protocols separate frequency channels are allotted to a user at the same time. In systems employing FHMA a technique of hopping over separate frequency channels is used.

A system architecture which is typically used is a cellular type architecture in which a plurality of base stations provide communication channels to many radio terminals. Each base station defines a cell and a radio terminal in a cell is capable of communicating with radio terminals in other cells or with radio terminals in the same cell, or with subscriber units in public services telephone networks (PSTN).

Typically, a radio terminal, associated with a subscriber unit, receives RF signals and processes them to provide the information required. When a signal is received at a radio terminal, the terminal has to acquire initial parameters of a control channel which enable it to find the precise frequency and the accurate timing of the control channel. Such procedures have to be performed on initialization only, and then other continuous algorithms are performed to maintain timing and frequency accuracies.

FHMA (frequency hopping multiple access) radio systems are described in US Patent 5,408,496 to Ritz et al.

In multiple access communication system

architectures, a timing and frequency synchronization has to be maintained at the base stations and at the subscriber units. Failure to maintain such synchronization may create collisions and misallotment of time slots and frequency slots which affect the quality of cellular communication.

In multiple access communication systems, the received signal level changes in time for various reasons. Various automatic gain control methods, which provide automatic gain control to the receiver amplifiers in order to maintain a constant signal level, are well known in the art.

The problem of providing hand-off between communication sectors with minimal disruption of service to the subscriber passing between sectors and other subscribers in his vicinity is well known.

Power control systems used in radio communication are well known in the art.

Power control systems and related apparatus are discussed in the following US Patents: 4,901,307 to Gilhousen; 5,056,109 to Gilhousen; and 5,257,283 to Gilhousen.

In multiple access communication systems, the time at which a signal is received the signal changes in time for various reasons, mostly the motion of mobile units. It is desirable in systems where time slots are assigned to particular transmissions to adjust the timing of transmissions so that transmissions fall within their assigned time slots.

When messages sent by a radio system need to be received correctly, existing systems typically require that an erroneously transmitted message be retransmitted in its entirety. Alternatively, existing systems which break messages into sub-messages typically require acknowledgement of each sub-message before transmission of the next sub-message.

In multiple access communication system architectures, especially when many users in a cell communicate simultaneously with users in the cell and with users in other cells, a group of frequencies may be allotted to more than one user at a time thus generating collisions. Typically, when a collision occurs communication between users having allotted the same group of frequencies is disrupted or even disconnected.

In order to address the above problems, sectorized radio communication systems are divided into a plurality of sectors, with each sector usually being served by its own antenna and each sector being assigned a set of channels. Such a prior art system is shown in Fig. 93. In Fig. 93, a base station 11050 serves three sectors 11060, 11070, and 11080.

Generally, near the foot of the base station of such a system sectorization, that is, distinguishing between transmissions belonging to different sectors, is problematic because within the vicinity of the foot of the base station, the base station loses the capability to isolate transmissions originating from different sectors. The cause of this problem, as will be appreciated from Fig. 93, is the close physical proximity between different sectors at the foot of the base station 11050 with all sectors coming together at the base station 11050.

Known solutions in the prior art involve the assignment of non-conflicting channels to the different sectors, particularly to adjacent or nearby sectors, in order to allow distinguishing between transmissions from different sectors based on the channel.

As is well known, a frequency channel in a wireless communication channel is subject to many sources of degradation. Thus, communication signals will not

always be communicated properly on a frequency channel. When operating a wireless communication system, it is desirable to be able to determine the "state" of a frequency channel in order to determine the likelihood of acceptable communication on the frequency channel. While channel estimators are known in the art, none adequately performs the tasks of determining the state of a frequency hopping channel. Another shortcoming of existing communication systems is found in the usage of channel state information.

Thus, apparatus and method for determining the state of a frequency channel are needed. Further, apparatus and method for using the channel state information to control the operation of the communication system to achieve improved communication are also needed.

Trunked radio services known in the art include the following: Special Mobile Radio (SMR) used in the United States; Public Mobile Radio (PMR) and Public Access Mobile Radio (PAMR) used in Europe and TETRA, described in ETSI standard ETS 300 396 / TETRA Direct Mode.

The disclosures of the above publications and of the publications cited therein are hereby incorporated by reference. The disclosures of all publications mentioned in this specification and of the publications cited therein are hereby incorporated by reference.

SUMMARY OF THE INVENTION

The present invention generally relates to method and apparatus for providing multiple access communications. In one aspect of the present invention, method and apparatus for handing off communications between a communicating party and a mobile communication radio which is moving from a first sector to an adjacent sector in a communication base station having multiple sectors is provided. In accordance with this aspect of the present invention, a mobile radio detects synchronization information from the first sector and searches for synchronization information from sectors adjacent to the first sector. The mobile radio determines when the synchronization information from the adjacent sector has stronger reception then the synchronization information from the first sector and then requests a hand off from the base station. The base station enables a three way communication link between the mobile radio in the first sector, the radio in the adjacent sector and the communicating party. Then, voice activity from the base station to the mobile radio is monitored and downlink communications from the base station to the mobile radio are handed off from the first sector to the adjacent sector when voice inactivity is detected. Also, voice activity from the mobile radio to the base station is monitored and uplink communications from the mobile radio to the base station are handed off from the first sector to the adjacent sector when voice inactivity is detected.

It is an object of the present invention to provide a radio communication system.

It is another object of the present invention to provide a system and process of providing a frequency hopping radio communication system.

It is a further object of the present invention to provide a system and method of providing a time hopping radio communication system.

Another object is to provide apparatus and method for handling subscriber units attempting to communicate in fringe areas.

Yet another object is to provide apparatus and method for providing service to subscriber units located at the foot of the base station.

Yet a further object is to provide apparatus and method of measuring the quality of service provided by a frequency hopping communication system.

It is also an object to provide apparatus and method that provide talk around capabilities.

It is another object to provide apparatus and method that acquire and track the communication channels of the present invention.

Another object of the present invention is to provide method and apparatus for handing off communications between sectors.

It is a further object to provide apparatus and method of automatic frequency control.

It is also an object to provide apparatus and method that provide linear power amplification of the signals to be transmitted.

Another object is to provide apparatus and method for automatic gain control.

Another object is to provide apparatus and method for determining and using channel state information.

These and other objects are further described in the description of the preferred embodiment that follows.

The present invention also seeks to provide a system and methods for acquiring a precise initial frequency and initial timing of a control channel

received at a subscriber unit in a frequency hopping multiple access communication system.

The present invention also seeks to provide methods for maintaining synchronization between a timing system in a subscriber unit and a timing system in a base station in a Frequency Hopping Multiple Access communication system (FHMA).

The present invention also seeks to provide a Frequency Hopping Multiple Access communication (FHMA) system having, at each subscriber unit, automatic frequency control for measuring and correcting frequency inaccuracies due to inaccuracies in frequency sources and Doppler shift effects at the subscriber units in order to improve detection of information transmitted via the communication systems.

It is another object of the present invention to provide improved methods for automatic frequency control.

The present invention also seeks to provide an improved automatic gain control system for use in a slotted radio communication system.

The present invention also seeks to provide an improved system for hand-off between sectors and a method for substantially seamless transfer of a mobile subscriber unit between sectors.

The present invention also seeks to provide an improved power control system for use in radio communication systems.

The present invention also seeks to provide time alignment in a radio communication system.

The present invention also seeks to provide an improved method and apparatus for processing received messages in order to reduce repeat transmissions of messages.

The present invention also seeks to provide methods for preventing collisions and for coordinating

channels between subscriber units in a Frequency Hopping Multiple Access (FHMA) communication system.

The present invention also seeks to provide a method for two subscriber units in an FHMA radio system to communicate directly with one another without use of a base station.

The solutions known in the prior art to the problem of sectorization in a vicinity of a base station have a drawback in that the assigning of non-conflicting channels places constraints on channel usage and requires a greater total number of channels than would otherwise be necessary. A solution which allows, at least in part, the assignment of conflicting channels to the different sectors, even to neighboring sectors, is thus desirable.

The present invention seeks to provide improved communication in a vicinity of a base station of a sectorized radio communication system.

There is thus provided in accordance with a preferred embodiment of the present invention a frequency-controlling signal receiving system including:

a frequency offset estimating modem operative to generate an estimated frequency offset value for a signal received by the modem; and

a local oscillator operative to receive the estimated frequency offset value and to generate a frequency converter control signal which is operative to cancel the estimated frequency offset.

Additionally, the system includes a frequency converter operative to receive the frequency converter control signal.

Preferably, the frequency offset estimating modem includes:

apparatus for receiving a signal with modulation and for canceling the modulation of the signal, thereby to generate a carrier wave;

an FFT (fast Fourier transform) computing

module for converting the waveform of the carrier wave from a time domain to a frequency domain, thereby to generate a final spectrum function in the frequency domain; and

a processor for computing a frequency offset by finding a frequency value which maximizes the final spectrum function.

The FFT computing module preferably includes:

a converter for converting the waveform of the carrier wave from a time domain to a frequency domain, portion by portion for a plurality of portions of the waveform, thereby to generate a plurality of intermediate spectrum functions; and

spectrum function combining unit operative to combine the intermediate spectrum functions to generate the final spectrum function.

Additionally, the spectrum function combining unit includes a summing unit for summing the spectrum function values for each of a multiplicity of frequencies.

In a preferred embodiment of the invention the carrier wave may include a distorted carrier wave.

There is also provided in accordance with a preferred embodiment of the present invention a method for self-synchronizing a signal including:

detecting a synchronization code embedded in a received signal and generating timing information associated with the detected synchronization code;

providing a local timing system; and

receiving the timing information and synchronizing the local timing system in accordance with the timing information.

Preferably, the detecting includes:

storing a representation of the synchronization code; and

correlating between the received signal and the

representation of the synchronization code.

Yet preferably, the correlating includes:

performing a sliding correlation operation within a window in which at least one element of the synchronization code is known to appear; and

finding an optimal correlation within the window,

wherein the timing information includes an indication of a time at which the optimal correlation appears.

Alternatively or additionally, the correlating may include:

performing a sliding correlation operation within a window in which at least N elements of the synchronization code are known to appear; and

finding M optimal correlations within the window,

wherein the timing information includes an indication of a time at which a predetermined one from among the M optimal correlations appears.

In accordance with a preferred embodiment of the present invention K < = M optimal correlations may correspond to K points in time and the method also includes verifying the K optimal correlations by comparing the K points in time to a known timing pattern of the K optimal correlations, wherein the K optimal correlations form a subset of the M optimal correlations.

Preferably, the received signal includes a synchronization code period and the synchronization code is embedded in the synchronization code period.

Further in accordance with a preferred embodiment of the present invention the received signal includes a super frame, wherein a plurality of synchronization code periods are embedded in the super frame and at least one synchronization code is embedded within each of the plurality of synchronization code

periods. Preferably, a synchronization code label is associated with each synchronization code, and timing information associated with the super frame is generated and is at least in part based on the plurality of the synchronization code labels.

There is also provided in accordance with another preferred embodiment of the present invention a method of maintaining a timing system in a subscriber unit in synchronization with a timing signal received from a base station in a frequency hopping multiple access communication system wherein a multiplicity of base stations communicate with a multiplicity of subscriber units over a frequency hopping multiple access communication network at a plurality of radio frequencies, the method including:

transmitting a timing synchronization signal at a selected slot over a control channel in the frequency hopping multiple access communication network to the subscriber unit;

receiving, over the control channel, the selected slot including the timing synchronization signal;

correlating the timing synchronization signal with a subscriber synchronization signal to provide an early correlation value and a late correlation value respectively;

generating a normalized difference between the early correlation value and the late correlation value to determine a time delay;

filtering the normalized difference with an infinite impulse response filter to provide a smooth response signal; and

providing the smooth response signal to a number controlled delay generator to generate a controlled delay correction signal.

Additionally, the method includes accumulating

a plurality of sequential smooth response signals smaller than a time increment at the number controlled delay generator.

Preferably, the method includes applying the controlled delay correction signal to modify the subscriber synchronization signal by a time increment substantially equal to 3 microseconds.

There is also provided in accordance with another preferred embodiment of the present invention a method of maintaining a timing system in a base station in synchronization with a timing signal received from a subscriber unit in a frequency hopping multiple access communication system wherein a multiplicity of base stations communicate with a multiplicity of subscriber units over a frequency hopping multiple access communication network at a plurality of radio frequencies, the method including:

transmitting a timing synchronization signal at a selected slot over a control channel in the frequency hopping multiple access communication network to the base station;

receiving, over the control channel, the selected slot including the timing synchronization signal;

correlating the timing synchronization signal with a base station synchronization signal to provide an early correlation value and a late correlation value respectively;

generating a normalized difference between the early correlation value and the late correlation value to determine a time delay;

filtering the normalized difference with an infinite impulse response filter to provide a smooth response signal; and

providing the smooth response signal to a number controlled delay generator to generate a

controlled delay correction signal.

Additionally, the method includes accumulating a plurality of sequential smooth response signals smaller than a time increment at the number controlled delay generator.

Preferably, the method includes applying the controlled delay correction signal to modify the base station synchronization signal by a time increment substantially equal to 6 microseconds.

Further in accordance with a preferred embodiment of the present invention there is provided a delay locked loop method for controlling a local timing system of a receiver by tracking the local timing system of an IF signal generated by the receiver, the method including:

storing an ideal waveform of a synchronization code embedded in an output signal of an RF/IF converter in the receiver;

generating a control signal monotonically related to a difference between the timing of the IF signal generated by the receiver and the timing system of the local timing system; and

supplying the control signal to the local timing system,

wherein the step of generating a control signal includes:

detecting first and second synchronization codes embedded in the output signal of the RF/IF converter at times t1 and t2 respectively, where t1 is a time preceding the estimated time at which the synchronization code embedded in the output signal of the RF/IF converter is maximally correlated with the ideal waveform and t2 is a time following the estimated time;

performing a first correlation between the detected synchronization code at time t1 and the stored ideal waveform;

performing a second correlation between the detected synchronization code at time t2 and the stored ideal waveform; and

computing the difference between the first and second correlations and defining a control signal on the basis of the difference.

Preferably, the step of computing and defining includes filtering the difference between the correlations.

Additionally, the method includes one of extracting and inserting at least one clock period to a counter in the timing system according to the control signal.

The local timing system may be a timing system at a base station.

There is also provided in accordance with another preferred embodiment of the present invention a frequency hopping multiple access communication system including:

a frequency hopping multiple access communication network;

a multiplicity of base stations, at least some of which receive and transmit information at a plurality of radio frequencies over the frequency hopping multiple access communication network; and

a multiplicity of subscriber units, each receiving and transmitting information at a plurality of radio frequencies via the frequency hopping multiple access communication network, wherein each subscriber unit includes:

a frequency control unit operative to determine and to reduce inaccuracies in each separate frequency of a hopping signal received from at least one base station, to acceptable values.

Preferably, each subscriber unit in the system also includes:

two separate antennas for separately receiving signals to achieve space diversity; and

two separate receivers, respectively coupled to the two separate antennas, wherein each receiver is operable to determine a quality of reception of a corresponding received signal, and the frequency control unit is selectably operable on one of the corresponding received signal having the best quality reception.

There is also provided in accordance with a preferred embodiment of the present invention a method of reducing inaccuracies in frequencies of an incoming hopping signal received at a subscriber unit from at least one base station in a frequency hopping multiple access communication system wherein a multiplicity of base stations communicate with a multiplicity of subscriber units over a frequency hopping multiple access communication network at a plurality of radio frequencies, the method including:

correlating the incoming signal with a synchronization signal to provide a complex correlation signal COR_PEAK determining a frequency offset;

filtering the imaginary portion IM_COR_PEAK of the complex correlation signal to provide a smoothed frequency offset signal; and

converting the smoothed frequency offset signal to voltage for a determination of an error correction voltage signal HFCS which is applied to a voltage controlled oscillator.

Preferably, the filtering includes:

applying a first closed loop for integration and feedback of the IM_COR_PEAK signal to provide a preliminary filtered signal; and

applying a second closed loop including a limiter for integration and feedback of the preliminary filtered signal between a limited numerical range to provide a smoothed frequency offset signal.

In a preferred embodiment of the present invention the limited numerical range is the numerical range between 0 and 1.

In accordance with another preferred embodiment of the present invention there is provided a method for controlling a local frequency source of a receiver which supplies a signal of a given frequency to an RF/IF converter in response to a control signal supplied to the local frequency source, the method being operative to maintain a fixed output frequency of the RF/IF converter, the method including:

storing an ideal waveform of a synchronization code embedded in a current output signal of the RF/IF converter;

generating a control signal monotonically related to a difference between a current output frequency of the RF/IF converter and a desired output frequency thereof; and

supplying the control signal to the frequency source,

wherein the step of generating comprises:

detecting a synchronization code embedded in a current output signal of the RF/IF converter; and

performing a complex correlation between the detected synchronization code and the stored ideal waveform thereof and defining the control signal as the imaginary part of the result of the complex correlation.

Preferably, the step of generating a control signal includes filtering the control signal. Alternatively or additionally, the step of generating a control signal includes compensating for nonlinearity of operation of the local frequency source.

There is also provided in accordance with a preferred embodiment of the present invention a subscriber unit in a frequency hopping multiple access communication system, the subscriber unit including:

at least one antenna for accepting over-the-air RF signals;

at least one receiver, coupled to the at least one antenna, and operative to receive the RF signals and to provide an IF output of the signals;

a local frequency source, coupled to the receiver, and operative to provide a signal of a given frequency to the receiver in response to an input signal;

a memory for storing an ideal waveform of a synchronization code signal;

a processor, coupled to the local frequency source, to the at least one receiver and to the memory, and operative to determine a frequency offset between a synchronization signal embedded in the RF signals and the given frequency of the local frequency source; and

a frequency control unit operative to maintain a fixed output frequency of the receiver by employing the frequency offset to generate a control signal which controls the local frequency source.

There is also provided in accordance with another preferred embodiment of the present invention automatic gain control apparatus for use in a receiver of a slotted radio communication system, the apparatus including a sample processor receiving a plurality of samples each indicating a sampled power level of an input signal and operative to produce a processed input power signal indicating a power level of the input signal, and error determining apparatus receiving the processed input power signal and operative to produce a power error signal.

Further in accordance with a preferred embodiment of the present invention the apparatus includes control apparatus receiving the power error signal and producing a gain control signal, and a variable-gain amplifier operative to control gain of the input signal based, at least in part, on the gain control

signal.

Still further in accordance with a preferred embodiment of the present invention the plurality of samples includes a plurality of samples of a current time slot.

Additionally in accordance with a preferred embodiment of the present invention the current time slot includes a plurality of symbols and each of the plurality of samples is associated with one of the plurality of symbols.

Moreover in accordance with a preferred embodiment of the present invention the variable-gain amplifier is operative to control gain of the input signal of a time slot succeeding the current time slot.

Further in accordance with a preferred embodiment of the present invention the sample processor includes averaging apparatus operative to compute an average of the plurality of samples.

Still further in accordance with a preferred embodiment of the present invention the sample processor includes absolute value computation apparatus operative to compute an average of absolute values of the plurality of samples.

Additionally in accordance with a preferred embodiment of the present invention the error determining apparatus includes logarithmic scaling apparatus operative to compute a logarithmic function of the processed input power signal.

Moreover in accordance with a preferred embodiment of the present invention the error determining apparatus includes filtering apparatus.

Further in accordance with a preferred embodiment of the present invention the filtering apparatus includes a lead-lag filter.

There is also provided in accordance with another preferred embodiment of the present invention an

automatic gain control method for use in a receiver of a slotted TDMA communication system, the method including receiving a plurality of samples each indicating a sampled power level of an input signal and producing a processed input power signal indicating a power level of the input signal, and receiving the processed input power signal and producing a power error signal.

Further in accordance with a preferred embodiment of the present invention the method includes receiving the power error signal and producing a gain control signal, and controlling the gain of the input signal based, at least in part, on the gain control signal.

Still further in accordance with a preferred embodiment of the present invention the plurality of samples includes a plurality of samples of a current time slot.

Additionally in accordance with a preferred embodiment of the present invention the current time slot includes a plurality of symbols and each of the plurality of samples is associated with one of the plurality of symbols.

Moreover in accordance with a preferred embodiment of the present invention controlling includes controlling the gain of the input signal of a time slot succeeding the current time slot.

Further in accordance with a preferred embodiment of the present invention receiving a plurality of samples and producing a processed input power signal includes computing an average of the plurality of samples.

Still further in accordance with a preferred embodiment of the present invention receiving a plurality of samples and producing a processed input power signal includes computing an average of absolute values of the plurality of samples.

Additionally in accordance with a preferred embodiment of the present invention receiving the processed input power signal and producing a power error signal includes computing a logarithmic function of the processed input power signal.

Moreover in accordance with a preferred embodiment of the present invention receiving the processed input power signal and producing a power error signal includes filtering.

Further in accordance with a preferred embodiment of the present invention filtering includes filtering via a lead-lag filtering method.

There is also provided in accordance with another preferred embodiment of the present invention a method for controlling the amplification of a receiver amplifier disposed upstream of a receiver modem, thereby to maintain a fixed modem input amplitude for a wide range of receiver input levels, the method including providing a current demodulator input amplitude value which is related to the amplitude of the current modem input, converting the current demodulator input amplitude value into a logarithmic-like unit, subtracting a desired logarithmic unit level from the converted current modem input amplitude value, thereby to provide a logarithmic-like unit difference value, and providing a control signal to the amplifier based on the logarithmic-like unit difference value.

Further in accordance with a preferred embodiment of the present invention the step of providing a control signal includes filtering the logarithmic-like unit difference value.

Still further in accordance with a preferred embodiment of the present invention the step of providing a control signal includes compensating for nonlinearity of operation of the amplifier.

There is also provided in accordance with

another preferred embodiment of the present invention an automatic gain control method for use in a receiver of a slotted radio communication system, the method including receiving a plurality of samples each indicating a sampled power level of an input signal and producing a processed input power signal indicating a power level of the input signal, and receiving the processed input power signal and producing a power error signal.

There is also provided in accordance with another preferred embodiment of the present invention automatic gain control apparatus for use in a receiver of a time slotted TDMA communication system, the apparatus including a sample receiver operative to receive a plurality of samples each indicating a sampled power level of an input signal and operative to produce a processed input power signal indicating a power level of the input signal, and an input power signal receiver operative to receive the processed input power signal and produce a power error signal.

There is also provided in accordance with another preferred embodiment of the present invention apparatus for controlling the amplification of a receiver amplifier disposed upstream of a receiver modem, thereby maintaining a fixed modem input amplitude for a wide range of receiver inpnt levels, the apparatus including input value provider operative to provide a current demodulator input amplitude value which is related to the amplitude of the current modem input, an input value converter operative to convert the current demodulator input amplitude value into a logarithmic-like unit, a subtracter operative to subtract a desired logarithmic unit level from the converted current demodulator input amplitude value, thereby providing a logarithmic-like unit difference value, and a control-signal provider operative to provide a control signal to the amplifier based on the logarithmic-like unit difference value.

There is also provided in accordance with another preferred embodiment of the present invention a method for seamlessly transferring a mobile subscriber unit, having an uplink and a downlink, from a first sector served by a first sector radio having a first antenna to a second sector served by a second sector radio having a second antenna, the method including monitoring a mobile subscriber unit in order to detect when the mobile subscriber unit passes from the first sector to the second sector, and switching the uplink and the downlink of the mobile subscriber unit from the first antenna to the second antenna, turning off the uplink and the downlink of the first sector radio and turning on the uplink and the downlink of the second sector radio, all within a time period which is short enough to cause substantially seamless communication.

There is also provided in accordance with another preferred embodiment of the present invention a method for seamlessly transferring a mobile subscriber unit, having an uplink and a downlink, from a first sector served by a first sector radio having a first antenna to a second sector served by a second sector radio having a second antenna, the method including monitoring a mobile subscriber unit in order to detect when the mobile subscriber unit passes from the first sector to the second sector, switching the uplink of the mobile subscriber unit from the first antenna to the second antenna, turning off the uplink of the first sector radio and turning on the uplink of the second sector radio, all while the mobile subscriber unit is not transmitting, and switching the downlink of the mobile subscriber unit from the first antenna to the second antenna, turning off the downlink of the first sector radio and turning on the downlink of the second sector radio while the first sector radio is not transmitting.

Further in accordance with a preferred

embodiment of the present invention the method also includes adapting at least one sector radio-subscriber unit control process to the second sector radio.

Still further in accordance with a preferred embodiment of the present invention the at least one control process includes time aligning.

Additionally in accordance with a preferred embodiment of the present invention the at least one control process includes power control.

Moreover in accordance with a preferred embodiment of the present invention the at least one control process includes AGC (automatic gain control) of the mobile subscriber unit.

Further in accordance with a preferred embodiment of the present invention the at least one control process includes sector radio AGC.

Still further in accordance with a preferred embodiment of the present invention the at least one control process includes DLL (delay lock looping) of the mobile subscriber unit.

Additionally in accordance with a preferred embodiment of the present invention the at least one control process includes sector radio DLL.

Moreover in accordance with a preferred embodiment of the present invention the at least one control process includes automatic frequency control (AFC) of the mobile subscriber unit.

Further in accordance with a preferred embodiment of the present invention the switching step includes setting up a 3-way conference between the first and second sector radios and the mobile subscriber unit.

There is thus provided in accordance with a preferred embodiment of the present invention apparatus operative to seamlessly transfer a mobile subscriber unit, having an uplink and a downlink, from a first sector served by a first sector radio having a first

antenna to a second sector served by a second sector radio having a second antenna, the apparatus including a mobile subscriber unit monitor operative to monitor a mobile subscriber unit in order to detect when the mobile subscriber unit passes from the first sector to the second sector, and a link switch operative to switch the uplink and the downlink of the mobile subscriber unit from the first antenna to the second antenna, to turn off the uplink and the downlink of the first sector radio and to turn on the uplink and the downlink of the second sector radio, all within a time period which is short enough to cause substantially seamless communication.

There is also provided in accordance with a preferred embodiment of the present invention apparatus operative to seamlessly transfer a mobile subscriber unit, having an uplink and a downlink, from a first sector served by a first sector radio having a first antenna to a second sector served by a second sector radio having a second antenna, the apparatus including a mobile subscriber unit monitor operative to monitor a mobile subscriber unit in order to detect when the mobile subscriber unit passes from the first sector to the second sector, a link switch operative to switch the uplink of the mobile subscriber unit from the first antenna to the second antenna, to turn off the uplink of the first sector radio and to turn on the uplink of the second sector radio, all while the mobile subscriber unit is not transmitting, and a second link switch operative to switch the downlink of the mobile subscriber unit from the first antenna to the second antenna, to turn off the downlink of the first sector radio and to turn on the downlink of the second sector radio while the first sector radio is not transmitting.

There is also provided in accordance with another preferred embodiment of the present invention a power control method for use in a radio communication

system including a first station and a second station, the method including choosing an initial transmitted power level for the first station, and performing the following steps iteratively transmitting a first message from the first station to the second station, receiving the first message at the second station, detecting a received power level for the first message at the second station, comparing the received power level to a predetermined value, transmitting a second message from the second station to the first station, the second message including an indication of a difference between the received power level and the predetermined value, receiving the second message at the first station, and modifying the transmitted power level for the first station based, at least in part, on the second message.

Further in accordance with a preferred embodiment of the present invention the initial transmitted power level is a maximum power level.

Still further in accordance with a preferred embodiment of the present invention the comparing includes smoothing a signal representing the received power level.

Additionally in accordance with a preferred embodiment of the present invention the smoothing is based, at least in part, on a value of the indication of difference from at least one previous iteration of the comparing.

Moreover in accordance with a preferred embodiment of the present invention the modifying includes storing the indication of the difference from the second message, and choosing an increment for modifying the transmitted power level based, at least in part, on the indication of the difference from the second message and based, at least in part, on a stored indication of the difference from a previous iteration.

Further in accordance with a preferred

embodiment of the present invention the modifying includes choosing an increment for modifying the transmitted power level based, at least in part, on a stored threshold.

Still further in accordance with a preferred embodiment of the present invention the choosing an increment is also based, at least in part, on a stored threshold.

Additionally in accordance with a preferred embodiment of the present invention the stored threshold is between approximately 5dB and approximately 10db.

Further in accordance with a preferred embodiment of the present invention the first station includes a subscriber unit and the second station includes a base station.

There is also provided in accordance with another preferred embodiment of the present invention a power control method for use in a radio communication system including a first station and a second station, the method including determining a desired received power level at the first station, transmitting a signal from the first station to the second station, the signal including an indication of a nominal transmitted power level for the first station, receiving the signal at the second station, detecting a received power level of the signal at the second station, comparing the received power level to the nominal transmitted power level and computing the transmission loss, and determining the transmitted power level of the second station based, at least in part, on the desired received power level at the first station and based, at least in part, on the transmission loss.

Further in accordance with a preferred embodiment of the present invention the signal from the first station to the second station includes a control channel signal, and the control channel signal includes

the indication of the nominal transmitted power level.

Still further in accordance with a preferred embodiment of the present invention the comparing and computing includes computing the difference between the received power level and the nominal transmitted power level.

Additionally in accordance with a preferred embodiment of the present invention the determining includes computing the sum of the desired received power level at the first station and the transmission loss.

Moreover in accordance with a preferred embodiment of the present invention the method includes determining the received power level at the first station, comparing the received power level at the first station to a predetermined value and outputting a signal representing the difference between the received power level at the first station and the predetermined value, transmitting a second signal from the first station to the second station, the second signal including an indication of the difference between the received power level at the first station and the predetermined value, receiving the second signal at the second station, and modifying the transmitted power level for the second station based, at least in part, on the second signal.

Further in accordance with a preferred embodiment of the present invention the first station includes a base station and the second station includes a subscriber unit.

Still further in accordance with a preferred embodiment of the present invention the detecting includes computing a signal to noise ratio.

Additionally in accordance with a preferred embodiment of the present invention the detecting includes computing a bit energy to noise density ratio.

Moreover in accordance with a preferred embodiment of the present invention the detecting

includes computing a carrier to interference ratio.

Further in accordance with a preferred embodiment of the present invention the radio communication system includes a multiple access system.

Still further in accordance with a preferred embodiment of the present invention the radio communication system includes a frequency hopping system.

There is also provided in accordance with another preferred embodiment of the present invention apparatus for use in a radio communication system including a first station and a second station, the apparatus including a first station transmitter having an initial transmitted power level for the first station and operative to transmit a first message from the first station to the second station, a second station receiver operative to receive the first message at the second station, a power level detector operative to detect a received power level for the first message at the second station, a power level comparator operative to compare the received power level to a predetermined value, a second station transmitter operative to transmit a second message from the second station to the first station, the second message including an indication of a difference between the received power level and the predetermined value, a first station receiver operative to receive the second message at the first station, and a power level controller operative to modify the transmitted power level for the first station based, at least in part, on the second message.

There is also provided in accordance with another preferred embodiment of the present invention power control apparatus for use in a radio communication system including a first station and a second station, the first station having a desired received power level, the apparatus including a first station transmitter operative to transmit a signal from the first station to

the second station, the signal including an indication of a nominal transmitted power level for the first station, a second station receiver operative to receive the signal at the second station, a power level detector operative to measure a received power level of the signal at the second station, a power level comparator operative to compare the received power level to the nominal transmitted power level and to compute the transmission loss, and a power level controller operative to determine the transmitted power level for the second station based, at least in part, on the desired received power level at the first station and based, at least in part, on the transmission loss.

Further in accordance with a preferred embodiment of the present invention apparatus includes a multiple access system.

Still further in accordance with a preferred embodiment of the present invention the radio communication system includes a frequency hopping system.

There is also provided in accordance with another preferred embodiment of the present invention a method for controlling the transmission power of a local transmitter according to link conditions, the method including determining link conditions by monitoring at least one characteristic of a local receiver associated with the local transmitter, and computing a level of transmission power based on the link conditions.

Further in accordance with a preferred embodiment of the present invention the determining by monitoring step also includes monitoring the transmission power of a remote transmitter which is transmitting to the local receiver and monitoring the noise floor level of a remote receiver which is receiving from the local transmitter and determining link conditions based on the local receiver characteristic, the remote transmitter transmission power, and the remote receiver noise floor

level.

There is also provided in accordance with another preferred embodiment of the present invention a method for controlling the transmission power of a local transmitter according to link conditions, the method including determining link conditions based on an indication of a characteristic of a remote receiver which is receiving from the local transmitter, which indication is received from the remote transmitter associated with the remote receiver, and computing a level of transmission power based on the link conditions.

Further in accordance with a preferred embodiment of the present invention the step of link condition determining also includes monitoring a characteristic of a local receiver associated with the local transmitter, initially generating an evaluation of link conditions on the basis of the local receiver characteristic, and improving the evaluation of link conditions upon receipt of the indication of remote receiver indication.

Still further in accordance with a preferred embodiment of the present invention the step of computing also includes comparing the initially generated link conditions evaluation to the improved link conditions evaluation and taking into account the result of the comparing step when subsequently performing the initial generating step.

Additionally in accordance with a preferred embodiment of the present invention the method include storing the result of the comparing step, thereby to take into account the result of the comparing step when performing more than one subsequent initial generating steps.

Moreover in accordance with a preferred embodiment of the present invention the receiver characteristic includes reception power.

Further in accordance with a preferred embodiment of the present invention the receiver characteristic includes the receiver's SNR (signal to noise ratio).

Still further in accordance with a preferred embodiment of the present invention the receiver characteristic includes the receiver's SIR (signal interference ratio).

Additionally in accordance with a preferred embodiment of the present invention the receiver characteristic includes the receiver's voice frame error rate.

There is also provided in accordance with another preferred embodiment of the present invention power control apparatus for use in a radio communication system including a first station and a second station, the apparatus including an initial power level chooser operative to choose an initial transmitted power level for the first station, and an iteration controller operative to control the iterative performance of the following apparatus a first message transmitter operative to transmit a first message from the first station to the second station, a first message receiver operative to receive the first message at the second station, a received power level detector operative to detect a received power level for the first message at the second station, a received power level comparator operative to compare the received power level to a predetermined value, a second message transmitter operative to transmit a second message from the second station to the first station, the second message including an indication of a difference between the received power level and the predetermined value, a second message receiver operative to receive the second message at the first station, and a transmitted power level modifier operative to modify the transmitted power level for the first station based, at

least in part, on the second message.

There is also provided in accordance with another preferred embodiment of the present invention power control apparatus for use in a radio communication system including a first station and a second station, the apparatus including a desired power level determinator operative to determine a desired received power level at the first station, a signal transmitter operative to transmit a signal from the first station to the second station, the signal including an indication of a nominal transmitted power level for the first station, a signal receiver operative to receive the signal at the second station, a received power level detector operative to detect a received power level of the signal at the second station, a received power level comparator operative to compare the received power level to the nominal transmitted power level and to compute the transmission loss, and a transmitted power level determinator operative to determine the transmitted power level of the second station based, at least in part, on the desired received power level at the first station and based, at least in part, on the transmission loss.

There is also provided in accordance with another preferred embodiment of the present invention a power control method for use in a radio communication system including a first station and a second station, the method including transmitting an initial transmitted power level message from the first station to the second station, receiving the first message at the second station, detecting a received power level for the first message at the second station, comparing the received power level to a predetermined value, transmitting a second message from the second station to the first station, the second message including an indication of a difference between the received power level and the predetermined value, receiving the second message at the the

first station, and modifying the transmitted power level for the first station based, at least in part, on the second message.

There is also provided in accordance with another preferred embodiment of the present invention a power control method for use in a radio communication system including a first station and a second station, the first station having a desired received power level, the method including transmitting a signal from the first station to the second station, the signal including an indication of a nominal transmitted power level for the first station, receiving the signal at the second station, detecting operative to measure a received power level of the signal at the second station, comparing the received power level to the nominal transmitted power level and to compute the transmission loss, and determining the transmitted power level for the second station based, at least in part, on the desired received power level at the first station and based, at least in part, on the transmission loss.

There is also provided in accordance with another preferred embodiment of the present invention apparatus for controlling the transmission power of a local transmitter according to link conditions, the apparatus including a link condition determinator operative to determine link conditions by monitoring at least one characteristic of a local receiver associated with the local transmitter, and a transmission power level computer operative to compute a level of transmission power based on the link conditions.

There is also provided in accordance with another preferred embodiment of the present invention apparatus for controlling the transmission power of a local transmitter according to link conditions, the apparatus including a link condition determinator operative to determine link conditions based on an

indication of a characteristic of a remote receiver which is receiving from the local transmitter, which indication is received from the remote transmitter associated with the remote receiver, and a transmission power level computer operative to compute a level of transmission power based on the link conditions.

There is also provided in accordance with another preferred embodiment of the present invention a method for time alignment of messages in a radio communication system having a first station and a second station, the method including determining a time alignment error of a message sent by the first station and received by the second station, sending a time alignment adjustment message from the second station to the first station, the time alignment adjustment message including a signal indicating the time alignment error of the message sent by the first station, and adjusting the timing of subsequent messages sent by the first station to the second station based, at least in part, on the time alignment adjustment message.

Further in accordance with a preferred embodiment of the present invention, determining includes comparing the time alignment error to a minimum error, and the sending and adjusting are performed only if the time alignment error is greater in magnitude than the minimum error.

Still further in accordance with a preferred embodiment of the present invention, the message sent by the first station includes a station identification, and determining includes comparing the station identification to a stored station identification of the first station, and sending and adjusting are performed only if the station identification matches the stored station identification.

Further in accordance with a preferred embodiment of the present invention, the message sent by

the first station includes a message type identification, and the time alignment adjustment message includes the message type identification.

Still further in accordance with a preferred embodiment of the present invention, adjusting is performed only if the message type identification matches a stored message type identification corresponding to a previous message sent by the first station.

Additionally in accordance with a preferred embodiment of the present invention, adjusting includes incrementally adjusting the timing of the subsequent messages based, at least in part, on a maximum adjustment for each of the subsequent messages.

Further in accordance with a preferred embodiment of the present invention, the subsequent messages include messages each having a message type and wherein the maximum adjustment for each one of the subsequent messages is based, at least in part, on the message type of the each one of the subsequent messages.

Still further in accordance with a preferred embodiment of the present invention, the method also includes determining the distance between the first station and the second station based, at least in part, on the time alignment error.

Further in accordance with a preferred embodiment of the present invention, the method includes storing time alignment errors of each of a plurality of messages, and determining the distance between the first station and the second station based, at least in part, on the stored time alignment errors.

Also provided, in accordance with a preferred embodiment of the present invention, is apparatus for time alignment of messages in a radio communication system having a first station and a second station, the apparatus including a time alignment determiner operative to determine a time alignment error of a message sent by

the first station and received by the second station, a message transmitter operative to send a time alignment adjustment message from the second station to the first station, the time alignment adjustment message including a signal indicating the time alignment error of the message sent by the first station, and a time alignment adjustor operative to adjust the timing of subsequent messages sent by the first station to the second station based, at least in part, on the time alignment adjustment message.

Further in accordance with a preferred embodiment of the present invention, the time alignment determiner includes a time alignment comparator operative to compare the time alignment error to a minimum error, and the message transmitter sends the time alignment adjustment message and the time alignment adjustor adjusts the timing only if the time alignment error is greater in magnitude than the minimum error.

Still further in accordance with a preferred embodiment of the present invention, the message sent by the first station includes a station identification, and the time alignment determiner includes a station identification comparator operative to compare the station identification to a stored station identification of the first station, and the message transmitter sends the time alignment adjustment message and the time alignment adjustor adjusts the timing only if the station identification matches the stored station identification.

Additionally in accordance with a preferred embodiment of the present invention, the message sent by the first station includes a message type identification, and the time alignment adjustment message includes the message type identification.

Still further in accordance with a preferred embodiment of the present invention, the time alignment

adjustor adjusts the timing only if the message type identification matches a stored message type identification corresponding to a previous message sent by the first station.

Yet further in accordance with a preferred embodiment of the present invention the alignment adjustor is operative to adjust the timing of the subsequent messages incrementally based, at least in part, on a maximum adjustment for each of the subsequent messages.

Also in accordance with a preferred embodiment of the present invention, the subsequent messages include messages each having a message type and the maximum adjustment for each one of the subsequent messages is based, at least in part, on the message type of the each one of the subsequent messages.

Also provided, in accordance with a preferred embodiment of the present invention, is a range determiner operative to determine the distance between the first station and the second station based, at least in part, on the time alignment error.

Further in accordance with a preferred embodiment of the present invention, the apparatus also includes a time alignment memory operative to store time alignment errors of each of a plurality of messages, and a range determiner operative to determine the distance between the first station and the second station based, at least in part, on the stored time alignment errors.

Also provided, in accordance with a preferred embodiment of the present invention, is apparatus for time alignment of uplink transmissions and including a plurality of time aligning subscriber units, and a time alignment determining base station operative to determine the time at which each uplink transmission arrives at the base station from a corresponding subscriber unit, to compute the timing error of each uplink transmission by

comparing the time of arrival thereof to a desired time of arrival, and to transmit each timing error to the corresponding subscriber unit, wherein each time aligning subscriber unit is operative to align its timing in order to reduce its timing error.

Also provided, in accordance with a preferred embodiment of the present invention, is a method for time alignment of uplink transmissions arriving from a plurality of subscriber units, the method including determining the time at which each uplink transmission arrives at a base station, computing the timing error of each uplink transmission by comparing the time of arrival thereof to a desired time of arrival, transmitting each timing error to the subscriber unit corresponding thereto, and at each subscriber unit, aligning timing of a subsequent uplink transmission in order to reduce its timing error.

Further in accordance with a preferred embodiment of the present invention, the timing alignment step includes performing a plurality of partial timing aligning substeps so as to gradually reduce the timing error.

There is also provided, in accordance with another preferred embodiment of the present invention, a method for processing received messages in order to reduce repeat transmissions of messages, the method including transmitting a message including a plurality of sub-messages, requesting retransmission of at least one incorrectly received sub-message, and if at least one previously incorrectly received sub-message is correctly received by retransmission, replacing at least one previously incorrectly received sub-message with its corresponding correctly received sub-message.

Further in accordance with a preferred embodiment of the present invention, the transmitting step includes transmitting via a frequency hopping

multiple access (FHMA) communication system.

Still further in accordance with a preferred embodiment of the present invention, the method also includes repeating the retransmission requesting step and the if-replacing step until all sub-messages have been correctly received.

Additionally in accordance with a preferred embodiment of the present invention, the method also includes marking each incorrectly received sub-message.

Further in accordance with a preferred embodiment of the present invention, the method also includes the step of, prior to transmitting a sub-message, providing error detection code for the sub-message individually, and wherein the error detection code is decoded upon receipt of the sub-message to determine whether or not the sub-message is received correctly.

Still further in accordance with a preferred embodiment of the present invention, the retransmission requesting step includes requesting retransmission of the entire message if at least one sub-message is received incorrectly.

Additionally in accordance with a preferred embodiment of the present invention, the step of requesting retransmission includes sending an acknowledgement when retransmission is no longer necessary, thereby to cause retransmission in the absence of an acknowledgement.

Further in accordance with a preferred embodiment of the present invention, the method also includes comparing at least first and second correct transmissions of the same sub-message when the sub-message is received correctly more than once.

Still further in accordance with a preferred embodiment of the present invention, the method, if the first and second correct transmissions are different,

also includes treating the more than once correctly received sub-message as an incorrectly received sub-message until a predetermined stopping criterion is reached.

Further in accordance with a preferred embodiment of the present invention, if the first and second correct transmissions are different, the predetermined stopping criterion includes receipt of a plurality of correct and identical transmissions of the sub-message.

Additionally in accordance with a preferred embodiment of the present invention, the predetermined stopping criterion includes receipt of a majority of correct and identical transmissions of the sub-message.

Further in accordance with a preferred embodiment of the present invention, the method also includes performing a validity check of the message, and if the validity check fails, repeating the validity check on at least one combination of correct transmissions of the sub-messages contained in the message.

Still further in accordance with a preferred embodiment of the present invention, the if-validity check repeating step includes repeating the validity check on all combinations until one of the combinations yields a successful validity check and requesting retransmission of at least a portion of the message if none of the combinations yields a successful validity check.

Also provided, in accordance with a preferred embodiment of the present invention, is apparatus operative to process received messages in order to reduce repeat transmissions of messages, the apparatus including a message transmitter operative to transmit a message including a plurality of sub-messages, a retransmission requester operative to request retransmission of at least one incorrectly received sub-message, and a message

replacer, operative to, if at least one previously incorrectly received sub-message is correctly received by retransmission, replace at least one previously incorrectly received sub-message with its corresponding correctly received sub-message.

There is also provided in accordance with another preferred embodiment of the present invention a method of preventing collisions and coordinating channels between subscriber units in a frequency hopping multiple access communication system wherein a multiplicity of base stations communicate with a multiplicity of subscriber units over a frequency hopping multiple access communication network at a plurality of radio frequencies, the method including:

providing, at each subscriber unit, a plurality of frequency channels for transmitting and receiving information signals over slots defined in a time domain and in a frequency domain;

transmitting the information over a plurality of slots; and

allowing each subscriber unit to skip transmission of at least one slot selected in accordance with a predetermined sequence.

Additionally, the method also includes:

receiving, at each subscriber unit, the transmitted slots;

recognizing a non-transmitted slot in the slots received; and

building an inactive slot, to replace the non-transmitted slot, by including in the inactive slot a plurality of inactive symbols having imparted a confidence level zero in at least one of a random sequence and an ordered sequence.

The predetermined sequence may be transmitted to each subscriber unit over a control channel.

Preferably, the method includes applying a

minimum weight to the inactive symbols during processing of the slots.

There is also provided in accordance with a preferred embodiment of the present invention a communication method wherein a transmitter within a first sector is to transmit to a first subscriber within the first sector, the method including:

transmitting to the first subscriber if the first subscriber is located within a fringe area; and

otherwise, determining, for each of at least one time slot, whether or not to transmit from the transmitter to the first subscriber during the at least one time slot.

Preferably, the determining step includes:

determining, for at least one time slot, whether, during the time slot, there exists a problematic subscriber associated with a neighboring sector who is located within a fringe area between the first and neighboring sectors and who is subject to interference due to transmission from the transmitter to the subscriber; and

if no problematic subscriber exists, transmitting in the time slot.

In a preferred embodiment of the present invention the determining step includes:

defining, for each of a plurality of time segments, a partition of the time during which the transmission occurs, each time segment including at least one time slot, determining whether or not the number of time slots within the time segment in which transmission did not take place exceeds a threshold number of time slots; and

if the threshold is exceeded, transmitting in a current time slot.

Preferably, the determining step includes:

for each of a plurality of positions of a

sliding time window including n>1 time slots, determining whether or not the number of time slots within the sliding window, as currently positioned, in which transmission did not take place exceeds a threshold number of time slots; and

if the threshold is exceeded, transmitting in a current time slot.

The determining step may also include determining whether to transmit with a probability $p<1$ or whether not to transmit.

In a preferred embodiment of the invention the method also includes:

assigning time-slots to each of a plurality of subsectors within the first sector and to each of a plurality of subsectors within the neighboring sector, each sector including central and peripheral subsectors such that the same time-slot is assigned to a peripheral subsector in the first sector and to a central subsector in the neighboring sector; and

assigning more power to downlink transmissions to subscribers within the peripheral subsectors than to downlink transmissions to subscribers within the central subsectors.

Additionally, the method includes allocating an air resource to the transmitters within the first sector so as to reduce the maximum probability, over the transmitters within the first sector, of existence of a problematic subscriber.

The air resource may include one of TDMA (time division multiple access) time slots, FDMA (frequency division multiple access) channels frequencies and FHMA (frequency hopping multiple access) time/frequency sequences. Preferably, each time slot includes an active time slot.

Additionally, the method includes, prior to the determining step, the step of transmitting with a

probability 1 if the first subscriber is inside the fringe area.

There is also provided in accordance with a preferred embodiment of the present invention a communication method wherein a first subscriber within a first sector is to transmit to a base station, the method including:

transmitting to the base station if the first subscriber is not located within a fringe area; and

otherwise, determining, for each of at least one time slots, whether or not to transmit during the time slot.

Preferably, the determining step includes:

defining, for each of a plurality of time segments, a partition of the time during which the transmission occurs, each time segment including at least one time slot, determining whether or not the number of time slots within the time segment in which transmission did not take place exceeds a threshold number of the time slots; and

if the threshold is exceeded, transmitting in a current time slot.

The determining step may also include:

for each of a plurality of positions of a sliding time window including $n>1$ time slots, determining whether or not the number of time slots within the sliding window, as currently positioned, in which transmission did not take place exceeds a threshold number of time slots; and

if the threshold is exceeded, transmitting in a current time slot.

Preferably, the determining step includes determining whether to transmit with a probability $p<1$ or whether not to transmit.

There is also provided in accordance with another preferred embodiment of the present invention a

communications system including apparatus for generating a full-duplex FHMA (frequency hopping multiple access) communication channel between two subscribers, and apparatus for generating a half-duplex FHMA communication channel between two subscribers who are proximate to one another.

Further in accordance with a preferred embodiment of the present invention, the communications system includes apparatus for determining, for a given pair of subscribers, whether to generate full-duplex or half-duplex communication and for assigning the given pair of subscribers to the full-duplex channel generating apparatus or to the half-duplex channel generating apparatus, accordingly.

Still further in accordance with a preferred embodiment of the present invention, the apparatus for determining is operative at least partly on a basis of a criterion of the extent of proximity between the given pair of subscribers.

Additionally in accordance with a preferred embodiment of the present invention, the criterion of the extent of proximity includes a criterion of signal quality.

Moreover in accordance with a preferred embodiment of the present invention, the apparatus for determining is operative to assign a given pair of subscribers to the full-duplex channel generating apparatus whenever sufficient air resources are available.

Further in accordance with a preferred embodiment of the present invention, the apparatus for determining is operative to store information associating each subscriber with one of a plurality of talk-groups and at least one pair of subscribers associated with the same talk-group are assigned to the half-duplex channel generating apparatus.

Still further in accordance with a preferred embodiment of the present invention, the apparatus for full-duplex channel generating and the apparatus for half-duplex channel generating are located within a base station.

There is also provided in accordance with another preferred embodiment of the present invention a method for generating a half-duplex FHMA communication channel between two subscribers who are proximate to one another within a communications system, the method including transmitting a channel request from a first subscriber to a second subscriber, determining a first measure of signal quality of the channel request received at the second subscriber, sending a request acknowledgement from the second subscriber to the first subscriber, determining a second measure of signal quality of the request acknowledgement received by the first subscriber, and generating a half-duplex FHMA communication channel between the first subscriber and the second subscriber only if both the first measure of signal quality and the second measure of signal quality meet a predetermined criterion.

Further in accordance with a preferred embodiment of the present invention, the generating step includes reversing use of an uplink channel and a downlink channel in exactly one subscriber from among the first and second subscribers.

Still further in accordance with a preferred embodiment of the present invention, the generating step includes using exactly one of an uplink channel and a downlink channel for both transmission and reception in both of the first and second subscribers.

There is also provided in accordance with another preferred embodiment of the present invention a communications system including apparatus for generating a mediated FHMA (frequency hopping multiple access)

communication channel between two subscribers, and apparatus for generating a direct FHMA communication channel between two subscribers who are proximate to one another.

Further in accordance with a preferred embodiment of the present invention, the communications system includes apparatus for determining, for a given pair of subscribers, whether to generate mediated or direct communication and for assigning the given pair of subscribers to the mediated channel generating apparatus or to the direct channel generating apparatus, accordingly.

Still further in accordance with a preferred embodiment of the present invention the apparatus for determining is operative at least partly on a basis of a criterion of the extent of proximity between the given pair of subscribers.

Additionally in accordance with a preferred embodiment of the present invention, the criterion of the extent of proximity includes a criterion of signal quality.

Moreover in accordance with a preferred embodiment of the present invention the apparatus for determining is operative to assign a given pair of subscribers to the mediated channel generating apparatus whenever sufficient air resources are available.

Further in accordance with a preferred embodiment of the present invention the apparatus for determining is operative to store information associating each subscriber with one of a plurality of talk-groups and at least one pair of subscribers associated with the same talk-group are assigned to the direct channel generating apparatus.

Still further in accordance with a preferred embodiment of the present invention the apparatus for mediated channel generating and the apparatus for direct

channel generating are located within a base station.

Additionally in accordance with a preferred embodiment of the present invention the direct FHMA communication channel includes a full-duplex communication channel.

Moreover in accordance with a preferred embodiment of the present invention the direct FHMA communication channel includes a half-duplex communication channel.

There is also provided in accordance with another preferred embodiment of the present invention, a method for generating a direct FHMA communication channel between two subscribers who are proximate to one another within a communications system, the method including transmitting a channel request from a first subscriber to a second subscriber, determining a first measure of signal quality of the channel request received by the second subscriber, sending a request acknowledgement from the second subscriber to the first subscriber, determining a second measure of signal quality of the request acknowledgement received at the first subscriber, and generating a direct FHMA communication channel between the first subscriber and the second subscriber only if both the first measure of signal quality and the second measure of signal quality meet a predetermined criterion.

Further in accordance with a preferred embodiment of the present invention, the generating step includes choosing one of the first subscriber unit and the second subscriber unit, and reversing use of an uplink channel and a downlink channel in the chosen subscriber unit.

Still further in accordance with a preferred embodiment of the present invention, the generating step includes using exactly one of an uplink channel and a downlink channel for both transmission and reception in

both of the first and second subscribers.

Additionally in accordance with a preferred embodiment of the present invention, the direct FHMA communication channel includes a full-duplex communication channel.

Further in accordance with a preferred embodiment of the present invention the direct FHMA communication channel includes a half-duplex communication channel.

There is also provided in accordance with another preferred embodiment of the present invention a communications method including generating a full-duplex FHMA (frequency hopping multiple access) communication channel between two subscribers in a communications system and generating a half-duplex FHMA communication channel between two subscribers who are proximate to one another in a communications system.

There is also provided in accordance with another preferred embodiment of the present invention apparatus for generating a half-duplex FHMA communication channel between two subscribers who are proximate to one another within a communications system, and including a channel requester operative to transmit a channel request from a first subscriber to a second subscriber, signal quality determining apparatus operative to determine a first measure of signal quality of the channel request received at the second subscriber, a request acknowledger operative to send a request acknowledgement from the second subscriber to the first subscriber, second signal quality determining apparatus operative to determine a second measure of signal quality of the request acknowledgement received by the first subscriber, and a half-duplex FHMA generator operative to generate a half-duplex FHMA communication channel between the first subscriber and the second subscriber only if both the first measure of signal quality and the second measure of

signal quality meet a predetermined criterion.

There is also provided in accordance with another preferred embodiment of the present invention a communications method including generating a mediated FHMA (frequency hopping multiple access) communication channel and a direct FHMA communication channel between two subscribers who are proximate to one another.

There is also provided in accordance with another preferred embodiment of the present invention apparatus operative to generate a direct FHMA communication channel between two subscribers who are proximate to one another within a communications system, including a channel requester operative to transmit a channel request from a first subscriber to a second subscriber, signal quality determining apparatus operative to determine a first measure of signal quality of the channel request received by the second subscriber, a request acknowledger operative to send a request acknowledgement from the second subscriber to the first subscriber, second signal quality determining apparatus operative to determine a second measure of signal quality of the request acknowledgement received at the first subscriber, and a FHMA communication channel generator operative to generate a direct FHMA communication channel between the first subscriber and the second subscriber only if both the first measure of signal quality and the second measure of signal quality meet a predetermined criterion.

Further in accordance with a preferred embodiment of the present invention the apparatus for generating a half-duplex FHMA communication channel between two subscribers who are proximate to one another includes apparatus for generating a half-duplex FHMA communication channel between two subscribers who are geographically proximate to one another.

There is also provided, in accordance with

another preferred embodiment of the present invention, a method for reducing interference in a communication system between a plurality of sectors in a vicinity of a base station including a fringe area of at least two of the sectors, the method including assigning a surrounding area which surrounds the base station to an individual one of the plurality of communication system sectors, and allocating air resources to communication system subscriber units within the surrounding area, thereby to allow communication between at least one of the subscriber units and at least one second party.

Further in accordance with a preferred embodiment of the present invention, the communication system includes an FHMA (frequency hopping multiple access) communication system.

Also provided, in accordance with another preferred embodiment of the present invention, is base station antenna apparatus including a plurality of sector antennas disposed at a height H relative to the ground, and an auxiliary antenna disposed at a height $h < H$ relative to the ground having a radiation pattern which includes an entire azimuthal vicinity of the base station.

Further in accordance with a preferred embodiment of the present invention, the sector antennas are located directly above the auxiliary antenna.

Alternatively in accordance with a preferred embodiment of the present invention, the sector antennas are not located directly above the auxiliary antenna.

Additionally in accordance with a preferred embodiment of the present invention, the auxiliary antenna is fed by a dedicated transmitter.

Further in accordance with a preferred embodiment of the present invention, the auxiliary antenna includes an omnidirectional antenna.

Also provided, in accordance with a preferred

embodiment of the present invention, is apparatus operative to reduce interference in a communication system between a plurality of sectors in a vicinity of a base station including a fringe area of at least two of the sectors, the apparatus including a sector assigner operative to assign a surrounding area which surrounds the base station to an individual one of the plurality of sectors, and an air resource allocator operative to allocate air resources to subscriber units within the surrounding area, thereby allowing communication between at least one of the subscriber units and at least one second party.

Further provided, in accordance with a preferred embodiment of the present invention, is a base station antenna construction method including disposing a plurality of sector antennas at a height H relative to the ground, and disposing at a height $h < H$ relative to the ground an auxiliary antenna having a radiation pattern which includes an entire azimuthal vicinity of the base station.

In accordance with one aspect of a preferred embodiment of the present invention, channel state information is derived from a set of communication symbols received during an individual time slot, although any set of communication signals or even a single communication signal can be used. For each of the received communication symbols, QPSK modulated symbols are hard detected to determine the actual transmitted signal. Then, the in phase and quadrature components of the received communication symbols in the plane of the modulation points are determined. Next, the sum of the in phase components is determined and the sum of the absolute value of the quadrature components is determined. The channel state is then determined from the ratio of the sum of the in phase components to the sum of the absolute value of the quadrature components.

In accordance with another aspect of the present invention, method and apparatus for erasing communication signals in accordance with the channel state are provided. In accordance with the method, the channel state is determined when the communication signal is received and then signals are erased if the channel state is not better than a predetermined level.

In accordance with a further aspect of the present invention, method and apparatus for selecting one of two communication signals for processing when diversity signals are received. In accordance with the method, a first channel state is determined when the first of two diversity communication signals is received and a second channel state is determined when the second of the two diversity communication signals is received. Then, based on the values of the first and the second channel states, one of the two diversity communication signals is selected for further processing.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention will be understood and appreciated more fully from the following detailed description, taken in conjunction with the drawings in which:

Fig. 1 represents the communication system of the present invention;

Fig. 2 illustrates the preferred common air interface links between a base station and the subscriber units;

Fig. 3A illustrates the preferred rules of transmission for traffic, control and access channels in each of the three sectors of the preferred embodiment of the present communication system;

Fig. 3B shows one way of time and frequency hopping multiple inputs and Fig. 3C illustrates a sequence generator for generating the hopping sequences of the present invention;

Fig. 4 shows a preferred time slot format;

Fig. 5 illustrates the preferred timing offset between uplink and downlink transmissions;

Fig. 6 illustrates a preferred sync-label slot format;

Fig. 7 illustrates the transmission of the SLS;

Fig. 8 is a block diagram of a base station;

Fig. 9 is a block diagram of a sector unit in a base station;

Fig. 10 illustrates a block diagram of the interconnection of the frame processor and the transmit (Tx) processing unit;

Fig. 11 illustrates the HDLC interconnections between the frame processing unit, the Tx processing unit, the Rx processing unit and the sector computer;

Fig. 12A illustrates a frame processing unit and Fig. 12B illustrates the block diagram of a quad frame board in the frame processing unit;

Figs. 13A and 13B illustrate a Rx processing unit and a Tx processing unit, respectively.

Fig. 14 is a block diagram of a micro sector unit in a base station;

Fig. 15 illustrates a preferred embodiment of a subscriber unit;

Fig. 16 illustrates a preferred embodiment of the RF portion of the subscriber unit;

Fig. 17 illustrates a preferred method of synthesizing frequencies;

Fig. 18 illustrates a preferred embodiment of a modem of the subscriber unit;

Fig. 19 illustrates a preferred embodiment of an ASIC in the modem of the subscriber unit;

Fig. 20 illustrates a preferred controller in the subscriber unit;

Fig. 21 illustrates a preferred voice package processor (VPP) in the subscriber unit;

Figs. 22A and 22B illustrate a preferred service board found in the subscriber unit;

Figs. 23 and 24 show the preferred signal processing performed when transmitting and receiving signals on traffic channels, respectively;

Figs. 25 to 31 show various error coding schemes;

Figs. 32 to 38 illustrate various steps used in other processes of the present invention;

Fig. 39 is a simplified block diagram illustrating the overall structure of the software components of the system of Fig. 1;

Fig. 40 is a simplified block diagram illustrating the structure of element 700 of Fig. 39 in greater detail;

Fig. 41 is a simplified block diagram illustrating the structure of element 706 of Fig. 40 in greater detail;

Figs. 42A and 42B are simplified block diagrams illustrating the structure of element 708 of Fig. 40 in greater detail;

Figs. 43A and 43B are simplified block diagrams illustrating the structure of element 714 of Fig. 42 in greater detail;

Fig. 44 is a simplified block diagram illustrating the structure of element 724 of Fig. 42 in greater detail;

Fig. 45 is a simplified block diagram illustrating the structure of element 718 of Fig. 42 in greater detail;

Fig. 46 is a simplified block diagram illustrating the structure of element 720 of Fig. 42 in greater detail;

Fig. 47 is a simplified block diagram illustrating the structure of element 716 of Fig. 42 in greater detail;

Figs. 48A and 48B are simplified block diagrams illustrating the structure of element 702 of Fig. 39 in greater detail;

Figs. 49 - 51 are simplified block diagrams illustrating the structure of a microsite;

Fig. 52 is a simplified block diagram illustrating the structure of a remote sector;

Fig. 53 is a generalized block diagram illustration of a portion of a subscriber unit in a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 54 is a flow chart illustration describing the operation of frequency acquisition in a channel feature acquisition algorithm which is performed at the

subscriber unit of Fig. 53 and is operative in accordance with a preferred embodiment of the present invention;

Fig. 55 is a flow chart illustration describing the operation of timing acquisition in a channel feature acquisition algorithm which is performed at the subscriber unit of Fig. 53 and is operative in accordance with a preferred embodiment of the present invention;

Fig. 56 is a generalized block diagram illustration of a portion of a subscriber unit in a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 57 is a simplified illustration of the operation of a delay locked loop in a subscriber unit which forms part of a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 58 is a generalized block diagram illustration of a portion of a base station in a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 59 is a simplified illustration of the operation of a delay locked loop in a base station of a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 60 is a generalized block diagram illustration of a portion of a subscriber unit in a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 61 is a simplified illustration of the operation of automatic frequency control in a frequency hopping multiple access communication system constructed

and operative in accordance with a preferred embodiment of the present invention;

Fig. 62 is a simplified block diagram of gain control apparatus constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 63 is a simplified flowchart illustrating the operation of a portion of the apparatus of Fig. 62;

Fig. 64A is a simplified flowchart illustrating an initialization method for the apparatus of Fig. 62;

Figs. 64B and 64C are simplified flowchart illustrations useful in understanding the method of Appendix A;

Fig. 65 is a simplified flowchart illustration of subscriber unit operations during a hand-off process provided in accordance with a preferred embodiment of the present invention;

Fig. 66 is a simplified flowchart illustration of base station operations during a hand-off process provided in accordance with a preferred embodiment of the present invention;

Fig. 67 is a simplified flowchart illustration of subscriber unit operations during a hand-off process provided in accordance with an alternative preferred embodiment of the present invention;

Fig. 68 is a simplified flowchart illustration of base station operations during a hand-off process provided in accordance with an alternative preferred embodiment of the present invention;

Fig. 69 is a simplified block diagram of a power control system for use in a radio communication system, the power control system being constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 70A is a simplified electronic circuit diagram illustrating the operation of the apparatus of Fig. 69;

Fig. 70B is a simplified flowchart illustrating the operation of a preferred implementation of step 6203 of Fig. 70A;

Fig. 71 is a simplified flowchart illustrating the operation of step 6204 of Fig. 70A according to an alternative embodiment of the present invention;

Fig. 72 is a simplified block diagram of a power control system for use in a radio communication system, the power control system being constructed and operative in accordance with an alternative preferred embodiment of the present invention;

Fig. 73 is a simplified flowchart illustrating the operation of the apparatus of Fig. 72;

Fig. 74 is a simplified block diagram of a power control system for use in a radio communication system, the power control system being constructed and operative in accordance with another alternative preferred embodiment of the present invention;

Fig. 75 is a simplified flowchart illustrating a preferred method for operating the apparatus of Fig. 74;

Fig. 76 is a simplified partly pictorial, partly block diagram illustration of the time alignment of a radio communication system constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 77 is a simplified flowchart illustration of a method for time alignment in the radio communication system of Fig. 76;

Fig. 78 is a simplified partly-pictorial, partly block-diagram illustration of a radio communication system constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 79 is a simplified block diagram illustration of a preferred method for operating the

system of Fig. 78;

Fig. 80 is a simplified block diagram illustration of a preferred error detection method useful in conjunction with the method of Fig. 79;

Fig. 81 is a simplified block diagram illustration of another preferred error detection method useful in conjunction with the method of Fig. 79;

Fig. 82 is a generalized block diagram illustration of a portion of a subscriber unit in a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 83 is a generalized block diagram illustration of a portion of a base station in a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 84 is a flow chart illustration describing the operation of a collision avoidance and channel coordination algorithm employed in the apparatus of Fig. 83;

Fig. 85 is a flow chart illustration describing the operation of another collision avoidance and channel coordinating algorithm which is performed at a base station and is operative in accordance with a preferred embodiment of the present invention;

Fig. 86 is a flow chart illustration describing the operation of a collision avoidance and channel coordinating algorithm which is performed at a subscriber unit and is operative in accordance with a preferred embodiment of the present invention;

Fig. 87 is a simplified pictorial illustration of a communication system constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 88A is a simplified flowchart illustration

of a preferred method for establishing and maintaining a talk around link between two subscriber units of the system of Fig. 87;

Fig. 88B is a simplified flowchart illustration of a preferred implementation of the method of Fig. 88A;

Fig. 89 is a simplified flowchart illustration of a preferred method for implementing a link-establishing step of Fig. 88B;

Fig. 90 is a simplified flowchart illustration of an alternative preferred method for implementing a link-establishing step of Fig. 88B;

Fig. 91 is a simplified flowchart illustration of the base station side of an alternative preferred method for establishing and maintaining a talk around link between two subscriber units of the system of Fig. 87;

Fig. 92 is a simplified flowchart illustration of the subscriber unit side of an alternative preferred method for establishing and maintaining a talk around link between two subscriber units of the system of Fig. 87;

Fig. 93 is a simplified pictorial illustration of a prior art sectorized communication system;

Fig. 94 is a simplified pictorial illustration of a sectorized communication system including a region around the foot of a base station, constructed and operative in accordance with a preferred embodiment of the present invention;

Fig. 95A is a simplified pictorial illustration of the base station 11100 of Fig. 94;

Fig. 95B is a simplified pictorial illustration of an antenna coverage pattern of a preferred implementation of the FBS antenna 11125 of Fig. 95A;

Fig. 95C is a side view of the antenna coverage pattern of Fig. 95B;

Fig. 96 is a simplified pictorial illustration

of an alternative embodiment of the present invention, which a region 11115 around the foot of the base station is incorporated into a sector;

Fig. 97 is a simplified pictorial illustration of the base station 11100 of Fig. 95A;

Fig. 98 illustrates a wireless communication system in which the apparatus and method of the present invention is used;

Fig. 99 shows the steps used to determine channel state by using the in phase and quadrature components of a received communication signal;

Fig. 100 illustrate a modulation plane and the determination of channel state by using the in phase and quadrature components of received signals as in Fig. 1;

Fig. 101 illustrates the processing apparatus in the subscriber units used to determine channel state;

Fig. 102 illustrated the process in accordance with another aspect of the present invention wherein communication signals from one of two diversity channels are selected for processing;

Fig. 103 illustrates the process in accordance with one aspect of the present invention wherein communication signals are erased; and

Fig. 104 illustrates a metric decision zone.

Attached herewith are the following appendices which aid in the understanding and appreciation of one preferred embodiment of the invention shown and described herein:

Appendix A is a detailed description of a preferred implementation of the apparatus and method of Figs. 62 and 63; and

Appendix B is a detailed description of the initialization method of Fig. 64.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

Fig. 1 illustrates the system 1 of a preferred embodiment of the present invention. The system 1 includes a base station 10, a plurality of subscriber units 12, three sectors 14, 15 and 16, a microsite 18 and a remote site 20.

In accordance with the present invention, the base station 10 establishes a communication link between a user on one subscriber unit 12 and one or more other users on other subscriber units 12. The base station 10 can also establish connections between one or more subscriber unit 12 and the Public Switch Telephone Network (PSTN).

In accordance with a preferred embodiment of the present invention, the communication system 1 is divided into sectors. While Fig. 1 illustrates three sectors 14 to 16, the preferred number of sectors utilized will depend mainly on the geographic location of the communication system 1 and on the number of subscriber units 12 which the system 1 needs to support.

Common Air Interface

The communication links between the base station 10 and subscriber units 12 is referred to as the common air interface. Fig. 2 illustrates a set of preferred communication channels in the common air interface. As shown the communication channels include a plurality of traffic channels (TCHs), one or more control channels (CCHs) and one or more access channels (ACHs). All of these channels are present in each sector 14 to 16. The TCHs operate in the uplink (transmissions from subscriber units 12 to the base station 10) and in the downlink (transmissions from the base station 10 to the subscriber units 12). The CCHs and the ACHs, however, operate only in one direction -- the CCHs in the downlink and the ACHs in the uplink.

In the communications system 1 of the present

invention, these channels are transmitted over a plurality of carrier frequencies. The number of carrier frequencies (or channels) utilized by the system 1 depends mainly on the available frequency spectrum and on the loading of the system 1. Each carrier frequency is preferably reused in each sector 14 to 16.

In one embodiment, in each sector 14 to 16, the system 1 uses ten carrier frequencies to define ten ~~uplink channels and uses a different ten~~ carrier frequencies to define ten downlink channels. In this embodiment, each uplink and downlink channel -- whether TCH, CCH or ACH -- operates within a channel bandwidth allocation of 25kHz. Also, in this embodiment, the uplink band of frequencies is contiguous as is the downlink band of frequencies, although such operation is not necessary to the present invention. In each sector 14 to 16, nine out of the ten available uplink carrier frequencies are utilized to implement nine uplink TCHs and nine of the ten available downlink carrier frequencies are utilized to implement nine of the ten downlink TCH transmissions. In each sector 14 to 16, the remaining uplink carrier frequency is utilized to transmit a single ACH while the remaining downlink carrier frequency is utilized to transmit a single CCH.

Each of the channels illustrated in Fig. 2 -- the TCHs, the CCHs and the ACHs -- carries predefined information. The TCHs transmit voice information, data information and inband overhead control signals between the base station 10 and the subscriber units 12. The CCHs transmit timing and control signals from the base station 10 to the subscriber units 12. At least one CCH is preferably transmitted perpetually by each sector unit 14 to 16. The ACHs transmit status and operational requests from the subscriber units 12 to the associated sector unit 14 to 16 in the base station 10.

Referring initially to Fig. 3A, several

preferred channel transmission rules in the sectors 14 to 16 will be discussed. The first rule concerns the transmission of information over the TCHs. As previously described, in accordance with one preferred embodiment of the present invention, the TCHs are defined over nine uplink and nine downlink carrier frequencies. It is appreciated that the number of sectors and number frequencies is given by way of example only and is not meant to be limiting. The transmission of information over a TCH is preferably via frequency hopping so that at a first time, a first block of information is transmitted on a first carrier frequency while at a second time, a second block of information is transmitted on a second carrier frequency, and so on. The transmitted information, therefore, is preferably hopped from carrier frequency to carrier frequency.

It is also preferred that each TCH be constructed of a periodic sequence of fixed, continuous and non-overlapping time slots. In accordance with a preferred embodiment, there are three time slots, but a larger or smaller number can easily be implemented. Some of the slots will be active, meaning that there is a transmission during that slot. Other slots will be passive, meaning that there is no transmission during that slot.

Referring to Fig. 3B, an example of frequency and time hopping of inputs from multiple users by the base station 10 is illustrated. In Fig. 3B there are five users, namely users A to E. User A has four blocks of information -- A1, A2, A3 and A4 -- which are queued for transmission. Similarly, user B has four blocks of information -- B1, B2, B3 and B4 -- that are queued for transmission, user C has four blocks of information -- C1, C2, C3 and C4 -- that are queued for transmission, user D has four blocks of information -- D1, D2, D3 and D4 -- that are queued for transmission, and user E has

four blocks of information -- E1, E2, E3 and E4 -- that are queued for transmission.

Referring to the right side of Fig. 3B, the available channels -- in this case ten -- which are defined by the carrier frequencies f_1 to f_{10}, are illustrated. Each of the ten illustrated channels are constructed of continuous time slots. There are four periods of transmission (designated I, II, III and IV) illustrated and each transmission period is divided into three time slots A, B and C.

The queued information on the left side of Fig. 3B is processed by assigning each block of information from each user to a time slot and then to a carrier frequency. For example, user A's information blocks, A1, A2, A3 and A4 are assigned to the time slot IB and to the channel f_3, to the time slot IIC and to the channel f_6, to the time slot IIIA and to the channel f_9, and to the time slot IVB and to the channel f_2, respectively. The blocks of information to be transmitted, therefore, are time hopped between the available time slots. They are also frequency hopped between the available channels. The assignment to time slots and to channels is preferably made in accordance with certain rules which will be discussed later. The queued information blocks for users B, C, D and E are similarly time and frequency hopped, as illustrated.

The frequency hopping is preferably done in accordance with a predefined sequence, which can be modified as necessary. It is preferred to select the sets of hopping sequences in an individual sector, for example, sector 14, such that during a selected time slot each of the carrier frequencies f_1 to f_{10}, is only being used by a single channel. Stated another way, no two channels within a sector employ the same carrier frequency at the same time. Selection of hopping sequences in accordance with this rule eliminates

interference within the sector 14. This preferred transmission rule and the generation of the sequences is more fully described in co-pending United States application number 080,075, filed on 18 June 1993 (and in co-pending Israel application number 103,620, filed 3 November 1992) which is hereby incorporated herein by reference.

The second preferred transmission rule also relates to the transmission of TCHs. In accordance with the second rule the set of frequency hopping sequences in one sector are selected such that no channel in that set of sequences employs the same carrier frequency at the same time as more than a predetermined number of channels in another set of frequency hopping sequences in an adjacent sector. The predetermined number of channels is the minimum number of channels possible and in a preferred embodiment is one. This transmission is discussed in greater detail in co-pending United States application number 080,075, which has been incorporated herein by reference.

Referring to Fig. 3C, a preferred sequence generator is illustrated. A sequence is chosen for the first sector. The sequences for the second sector are generated by a left cyclic shift of each line of the sequences of the first sector by one location. The sequences for the third sector are also generated by a left cyclic shift of each line of the sequences of the second sector by one location.

Fig. 3A also describes the preferred transmission rules for CCHs which transmit timing and control signals from the base station 10 to the subscriber units 12. As already described, there is a single CCH in each sector 14 to 16. In accordance with a preferred embodiment, the single CCH is perpetually transmitted over one of the plurality of carrier frequencies. Therefore, the CCH is preferably not

frequency hopped. Further, the CCH in each sector 14 to 16 is preferably assigned one of the three time slots and is transmitted only in that time slot. Therefore, in a preferred embodiment, the CCH in sector 14 is transmitted during time slot A, the CCH in sector 15 is transmitted during time slot B and the CCH in sector 16 is transmitted during time slot C.

Fig. 3A also describes the preferred transmission rules for ACHs. In each sector 14 to 16, most of the ACHs are transmitted in a slotted ALOHA format. It is preferred to utilize a variant of the stabilized slotted. Typically, a portion of the ACHs is in a format similar to the CCH.

As already stated, each channel is constructed of continuous time slots. Fig. 4 illustrates a preferred format of an active time slot 20. The slot 20 covers 2.22 msec. and includes a total of 41 symbols. Each symbol, therefore, is 54 micro seconds long. Each symbol may consist of two bits of information. Two of the symbols, the first and last ones, are left inactive as guards to protect against time shifted time slots. When the time slot is active, therefore, the middle 39 symbols are active and used to transmit information.

It is preferred to transmit the uplink slots 21 from the subscriber unit 12 to the base station 10 offset in time from the transmission of downlink slots 22 from the base station 10 to the subscriber units 12, as shown in Fig. 5. The time offset is preferably 1.11 msec. Offsetting the uplink transmissions from the downlink transmissions allows the system of the present invention to be compatible with either full duplex (receive and transmit simultaneously) or half duplex operation.

As stated previously, timing and control information for the system 1 is transmitted over a CCH. One of the CCH messages contains synchronization information and is transmitted in sync/label slots (SLS).

Fig. 6 shows a preferred format of a SLS 23. The SLS 23 includes a sync 24 consisting of 20 symbols, a label 25 consisting of 19 symbols and guard symbols 26 and 27 at the front and the end of the SLS 23. The sync 24 is a deterministic synchronization code. The subscriber units 12 detect the sync 24 when it is transmitted and synchronize their operation accordingly. The label 25 is preferably constructed of eight symbols that indicate which communication site transmitted the SLS, four symbols indicating which sector within the site the subscriber unit is located in, six symbols indicating the index of the received SLS 23 within a superframe of information and 1 symbol indicating whether coherent or differential modulation is being utilized.

The transmission of the SLS 23 is illustrated in Fig. 7. As illustrated, each sector 14 to 16 is assigned its own time slot. The SLSs are transmitted on the average, every 72 time slots in a staggered manner so as to ensure the reception of at least one SLS in one's own operating time slot. The entire pattern has a period of 216 time slots. Information about the base station 10 is also transmitted on the CCH.

It is preferred to have two modes of demodulation operation. The first mode of operation is coherent demodulation and the second mode of operation is differential modulation. Coherent demodulation offers improved performance when the received signal has a stable carrier phase, but requires good synchronization to the phase of the received, yet unmodulated, signal. It is anticipated that some channel conditions -- such as severe fading -- may be such that coherent demodulation may be erratic. It is therefore preferred to utilize differential demodulation as well. The mode of operation will be initially selected for individual communication sites. The base station 10 instructs the subscriber units 12 which mode of operation is being used by setting

a bit in the SLS 23.

The Base Station

The main task performed by the base station 10 is to connect subscriber unit users 12 with a PSTN or with another subscriber unit 12. The base station 10 provides various voice services to the users, including basic telephony, voice mail, group dispatch calls and individual dispatch. The basic telephony services include incoming phone calls from a PSTN, outgoing phone calls outside the base station 10 area via a PSTN and phone calls between subscribers. The preferred features provided for telephony services includes speed dialing based on a phone number library stored in the subscriber unit 12, fleet phone numbers library, call waiting, call forwarding, emergency call, display of caller phone number, camp-on capability, prescribed telephony privileges per subscribe unit 12 and answering modes that include auto answer and ringing. The features of the group dispatch services includes emergency dispatch and camp-on capability on system resources and on group dispatch. Individual dispatch can be provided via full duplex and via half duplex.

The base station 10 also provides data services, including group messaging, individual messaging and virtual circuit. For individual messaging, basic individual messaging services, special delivery and registered delivery are provided. Also provided are: warnings to the sender; guaranteed delivery; unique identification of messages, a dropped message log file; a message receipt time stamp; two levels of message priority; delayed delivery; an address distribution list and an alternate destination address. The virtual circuit provides a connection oriented, packet switched type of service between two users. It is preferred to measure inactivity time for each virtual circuit. The

base station 10 also provides fax services. Another basic task performed by the base station 10 is to manage a date base of the subscriber unit 12 users.

A preferred embodiment of the base station 10 is illustrated in Fig. 8. The base station 10 includes a first sector unit 30, a second sector unit 31, a third sector unit 32, a microsector unit 34, a redundant sector 35, a PABX 36, a voice mail unit 38, a central frequency source unit 40, an administration computer 42, a central controller 44, a data base server 46, a local administrator computer 48, a terminal server 50, an ethernet local area network 52, a power supply 54 and data computers 55.

The sector units 30 to 32 establish the previously discussed communication channels within the sectors 14 to 16, respectively. The micro sector unit 34 establishes communication channels with the microsite 18. The redundant sector 35 provides redundant communication channels for the various sector units. Communication with the PSTN is provided via the PABX 36.

Referring to Fig. 9, a block diagram of a sector unit 30 to 32 is illustrated. The sector unit 30 includes antennae 56 and 57, antenna boxes 58 and 59, wideband amplifier units 60 and 61, each of which include an input combiner 62, a preamplifier 63, a power divider 64, amplifiers 65 and an output combiner 66, multicoupler units 67A and 67B, a receive (Rx) processing unit 68, a transmit (Tx) processing unit 70, a frame processing unit 72, a sector controller 74 and a power supply 76.

Both antennas 56 and 57 are utilized during a receive cycle and during a transmit cycle. During a transmit cycle, the antenna 57 transmits signals provided by the Tx processing unit 70 through the wideband amplifier 69 and the antenna box 58 into a sector while the antenna 56 transmits signals provided by the Tx processing unit 70 through the wideband amplifier 61 and

the antenna box 59. During a receive cycle, the antennae 56 and 57 both receive signals that are provided to the multicouplers 67A and 67B, respectively, before being sent to the Rx processing unit 68. Thus, two complete receive paths are established to provide spatially independent received signals. The antennae 56 and 57 are preferably placed more than several wavelengths from each other. The best received signal is selected from the two and then routed for frame processing.

Referring to Fig. 10, a block diagram of the interconnection between the frame processor unit 72 and the Tx processing unit 70 is illustrated. The frame processing unit preferably includes forty (40) FPUs, while the Tx processing unit 70 preferably includes ten (10) TPUs. Each FPU is connected by the sector controller to handle one conversation at a time. Thus, as illustrated, up to forty conversations can be handled by the forty FPUs. When an FPU is instructed to handle a conversation, the sector controller provides the FPU with a key that specifies the sequence of frequencies to be utilized. The key can also specify the time slot to be utilized. Each FPU is connected to each TPU in the Tx processing unit 70 through a modified HDLC bus. Each TPU provides signals to be transmitted on a predefined frequency (channel). The FPU sends the part of the conversation to be transmitted to the appropriate TPU on the HDLC bus in accordance with the key received from the sector controller. The time slot of transmission is determined in accordance with the time position of the signal to be transmitted on the HDLC bus. The FPU properly places the signal to be transmitted in the proper time sequence on the HDLC bus in accordance with the key provided by the sector controller.

Fig. 10 also represents the interconnection between the frame processor unit 72 and the Rx processing unit 68. The Rx processing unit has ten RPUs

(represented by the TPUs in Fig. 10), each of which receives signals from the appropriate sector on a predefined frequency (channel). The sector controller instructs the RPU which FPU should receive the signal through a key. The RPU then sends the information received in the proper timing on the HDLC bus to the appropriate RPU where the signal transmitted is reconstructed. Fig. 11 illustrates the HDLC interconnections in greater detail.

Referring to Fig. 12A, a preferred frame processing unit 72 is illustrated. The frame processing unit 72 includes a bridge interface board 80, a dual T1 board 82 and ten quad frame boards 84. The dual T1 board 82 provides a communication interface to the PABX 36. The bridge interface board 80 provides an interface to the HDLC bus. Each one of the quad frame boards 84 includes four FPUs so that a total of forty FPUs are provided as previously discussed.

The quad frame board 84 is illustrated in greater detail in Fig. 12B. Each board 84 includes a host CPU 86. Each board also includes four FPUs, each of which includes a digital signal processor 88, a vocoder 90, buffers 92 and 94, a decoding PAL 96, an oscillator 98, a test buffer 100, a boot ram 102, a viterbi decoder 104, data ram 106 and program ram 108. The host CPU 86 controls the communications on the HDLC bus as well as the overall functioning of each FPU. Each FPU processes the information to be transmitted or received. On the transmit side, the processing includes compressing the information with a vocoder, error correction coding and interleaving the signal. On the receive side, the processing includes the reverse steps.

Referring to Figs. 13A and 13B, the Rx processing unit 68 and Tx processing unit 70, respectively, are illustrated. The Rx processing unit 68 preferably includes bridge interface boards 110 and 112,

Rx RF/IF boards 114 and 116, digital Rx boards 118 and 120 and reference generator 121. The bridge interface boards 110 and 112 provide an interface to the Tx processing unit 70, where an interface to the frame processing unit 72 is provided. The signals being received are provided to the Rx RF/IF boards 114 and 116 by the multicouplers unit 67A and 67B, respectively, where the signals are converted to digital form. Once in digital form, the signals are provided to the digital Rx boards 118 and 120, where the signals are put in proper form for transmission to the frame processor 72.

Referring to Fig. 13B, the Tx processing unit 70 is similar to the Rx processing unit 68, but is configured to send signals in the opposite direction. The Tx processing unit 70 includes bridge interface boards 122 and 124, Tx RF/IF boards 126 and 128, digital Tx boards 130 and 132, and reference generator 133. The bridge interface boards 122 and 124 provide an interface for the Tx RF/IF boards 126 and 128 to the frame processing unit 68. The bridge interface boards 122 and 124 also provide an interface to the Rx processing unit 68, as shown. The digital Tx boards 126 and 128 collect the signals to be transmitted on the various channels and properly format these signals. The Tx RF/IF boards 126 and 128 convert the signals to be transmitted to analog form for transmission.

Referring back to Fig. 9, the sector controller 74 provides control of the various functions performed by a sector and is controlled by the central controller 44. Control and data signals are exchanged between the sector controller 74 and the central controller 44 via the LAN 52. In addition to controlling the processing that is performed by the sector 30 to 32, the sector controller 74 also keeps a log book of all the active users under supervision. The sector controller 74 provides this information to the central controller 44 via the LAN 52.

Referring back to Fig. 8, the description of the base station 10 will be completed. Each of the sector units 30 to 32 and the micro sector unit 34 is interfaced to the PABX 36. The PABX 36 provides the connection of the system 1 to the public switch telephone network (PSTN). The interface is via a standard 2Tx1 connection. The PABX 36 also provides three way conferencing, routing, least cost routing of long distance calls, voice mail interfacing, dispatch bridging, user services support and metering functions.

The voice mail unit 38 provides voice mail capability. The central frequency source unit 40 provides sector and timing references to the sector units 30 to 32 and to the micro sector unit 34. The frequency source is preferably generated by a rubidium atomic reference.

The administrative computer 42 tracks the configuration grouping, tracks administration activities, performs network management, performs bit management and performs the system initialization. It is preferably implemented with a Sun Sparcstation.

The central controller 44 provides various functions including call management, dispatch management, control of the PABX 36, voice mail interfacing, operational mode management, subscribers management, call management, billing information and reports generation. It is preferably implemented with a 486 PC compatible computer. The data base server 46 stores user data concerning user rights, status, calls and airtime. It also provides basic data base management and services to all data base clients, such as the local operator, fleet administrators and remote operators. The local administrator computer 48 provides maintenance and operational control of the base station 10. An ethernet local area network (LAN) 52 is provided to enable communications between the various components connected to the

network.

The micro sector unit 34 is illustrated in greater detail in Fig. 14. The micro sector unit 34 provides communication channels to the microsites 18. The micro sector 34 includes one or more microwave transceivers 134, a video processing unit 136, a micro Tx processing unit 138, a micro Rx processing unit 140, a slot selector unit 142, a frame processing unit 144, a sector controller 146 and a power supply 148. The receive and transmit signal processing is performed by the micro Tx processing unit 138, by the micro Rx processing unit 140, by the slot selector unit 142, by the frame processing unit 144 and by the sector controller 146. The processing steps performed by the micro sector 34 are the same as those performed by the sectors 30 to 32 as previously discussed.

Subscriber Unit

Fig. 15 is a block diagram of the subscriber unit (SU) 12 of the present invention. Functionally, the SU 12 preferably supports the same features supported by the base station 10. As previously discussed, these features include, but are not limited to, telephony interconnect, two-way radio, and data communications. The telephony interconnect features include incoming calls, outgoing calls, call waiting, call forwarding, call disconnect, call flip-flop, switching to dispatch and back while on call, dialing while conversing, last number redial and speed dialing. The two-way features include incoming dispatch and dispatch initiation.

As shown in Fig. 15, the SU 12 includes a first antenna 202, a second antenna 204, a radio unit (RU) 206, a baseband unit (BBU) 208, a service board (SB) 210, a GPS interface 211 and a man machine interface (MMI) 212. The RU 206 includes a duplexer 213, a receiver channel 214, a diversity receiver channel 216, a gain and frequency control unit 218, a transmitter 220, a synthe-

sizer 222 and a gain control unit 224. The BBU 208 includes a modem 226, a controller 228, a voice processing package (VPP) 230 and a MMI interface 232.

The two antennae 202 and 204 establish the previously discussed communication channels with the base station 10. They are preferably 14 inch, stainless steel, high performance collinear antennae which are magnetically secured to a vehicle. They preferably operate in the frequency range of 890 to 950 MHz, although tuning to any desired frequency is possible. The two antennae 202 and 204 are preferably omnidirectional and have linear vertical polarization, a free space gain of 3 dBi and a maximum VSWR of 1.5:1.

The SU 12, when transmitting to the base station 10, transmits only on the first antenna 202. When the SU 12 receives transmissions from the base station 10, however, both the first and second antenna 202 and 204 are utilized to achieve space diversity. As with the base station 10, the best signal is selected for processing.

Referring to Fig. 16, a more detailed block diagram of the transmitter 220, the receivers 214 and 216 and the synthesizer 222A and 222B is illustrated. The transmitter 220 received I and Q inputs from the modem 226. The I and Q inputs are amplified by amplifiers 234 and 236, respectively, and then filtered by low pass filters 238 and 240, respectively. The filtered I and Q signals are then modulated by a modulator 242. The modulator 242 is supplied with a hopping oscillator signal, TxLO, from the synthesizer 222A so that the signal transmitted by the transmitter 220 is frequency hopped.

The modulated signal is then controllably attenuated by an Up attenuator 244, filtered by a high pass filter 246, amplified by amplifiers 248 and 250, down attenuated by attenuator 252 and filtered by low

pass filter 254 before being supplied to the duplexer 213 for transmission by the antenna 202. As can be seen, the gain of the transmitted signal is controlled by the attenuators 244 and 252. The attenuator 244 is controlled by the signal LEVEL CONTROL which is received from the gain control unit 224. The gain control unit 224 receives its inputs (EN5, DATA and CLK) from the modem 226.

The antenna 202, in addition to transmitting signals from the transmitter 220 to the base station 10, also receives signals which are transmitted by the base station 10. Those received signals are transmitted through the duplexer 213 to the receive channel 214. The received signal is amplified with a low noise amplifier 256, filtered by a bandpass filter 258 and then downconverted by a mixer 260 to IF.

The received signal can be a frequency hopped signal. Where appropriate, therefore, the mixer 260 downconverts the received signal by mixing the received signal with an oscillating signal from the synthesizer 222B which is also hopping.

The downconverted signal is then filtered by a bandpass filter 262 and then amplified by a variable amplifier 264. The variable amplifier 264 is gain controlled in accordance with a signal, AGC, received from the gain & frequency control unit 218.

The signal transmitted from the base station 10 to the SU 12 is also received by the second antenna 204. The received signal is then processed by the receiver 216. First, the received signal is filtered by a band pass filter 272 and then it is processed by a low noise amplifier 274, by a filter 276, by a mixer 278, by a filter 280, by a gain controlled amplifier 282, by a mixer 284, by a filter 286 and by an amplifier 288 in a similar fashion to the processing performed in the receiver 214. Note, however, that gain control of the

diversity channel is accomplished by a separate control signal, AGC(D).

The synthesizer 222A and 222B generates the signals necessary to modulate the signal transmitted by the transmitter 220 and to downconvert the signal received by the receiver channels 214 and 216. A fixed frequency generator 296 generates a signal having a frequency of 14.4 MHz. This signal is provided to a transmit hopping synthesizer 294. The transmit hopping synthesizer 292 generates an output signal with a frequency that varies from 757.8 MHz to 797 MHz in accordance with control signals, TxHOP CONTROL, which are provided by the modem 226. The output of this synthesizer is output to a divider 294 where the frequency of the synthesized signal is divided by eight. Then the signal is filtered by high pass filter 296 before being supplied to one of the inputs of a mixer 298.

The other input to the mixer 298 is supplied by a fixed synthesizer 300 which synthesizes a frequency of 801.29375MHz in accordance with an input from an oscillator (TOXO) 302. The TOXO 302 is controlled by a control signal, AFC, which is generated by the modem 226. The control signal AFC is varied to keep the frequency of transmission constant. The output of the mixer 298 therefore, is a hopped frequency which is filtered by a bandpass filter 304 before being input to the modulator 242 where it is used to modulate the I and Q signals that have been provided by the modem 226.

An output from the reference 290 is also supplied to a receive (Rx) hopping synthesizer 306. This synthesizer 306 generates a signal that is hopped in the frequency range of 882.3 MHz to 902.2 MHz. The frequency hopping is controlled by control signal Rx HOP CONTROL generated by the modem 226. The output of the Rx hopping synthesizer 306 is supplied to a divider 308 and then to

a high pass filter 310 before being supplied to an input of a mixer 312. The other input of the mixer 312 is supplied by the fixed synthesizer 300. The output of the mixer 312 is filtered by a band pass filter 314 and results in a signal that ranges in frequency from 1021.86875 MHz to 1026.84375 MHz. The signal is then split into two signals by a splitter 316. One of the split signals is supplied to the downconverter 260 in the first receiver channel 214. The other signal from the splitter 316 is supplied to the downconverter 278 in the diversity channel 216.

The reference 290 also supplies a reference frequency to an IF synthesizer 318. The IF synthesizer 318 generates a signal having a frequency of 86.38875 MHz. This signal is split by a splitter 320 and then supplied to downconverters 266 and 284 in the receiver channels 214 and 216, respectively.

Referring to Fig. 17, the preferred method for generating the hopping oscillators is illustrated. A frequency reference 310 drives a numerical controlled oscillator 311. The two sine and cosine outputs (in quadrature) from the numerical controlled oscillator drive a single sideband (SSB) mixer 312. An additional RF input is provided from a voltage controlled oscillator 313 through a directional coupler 314 at frequency f_{out}. The output of the mixer 312 at frequency $f_{out} - f_{NCO}$ is down divided by N using a frequency divider 315. A phase lock loop, formed by VCO 313, SSB mixer 312, divider 315, phase detector 316, and loop filter 317, is used to lock the VCO onto an output frequency such that $f_R \div N = f_{out} - f_{NCO}$. Coarse frequency is set by f_R and N and f_{NCO} is used for fine and fast frequency hopping.

Fig. 18 illustrates the circuitry of the modem 226. The modem includes a digital to analog converter (DAC) 322, an analog to digital converter (ADC) 324, a converter interface (CNVR) 326, an ASIC 328 and a digital

signal processor 330. The DAC 322, during the transmit function, receives I and Q signals from the ASIC 328 and converts those signals to analog form before supplying them to the transmitter 222. The ADC 324, during the receive function, receives a signal IF from the receiver channel 214 and a diversity signal IF-d from the diversity receiver channel 216. It converts these signals to digital form and then supplies them to the ASIC 328 for further processing. The CNVR 326 provides the control signals for automatic gain control of the first receiver channel 214 (AGC), for automatic gain control of the diversity receiver channel 216 (AGC-d) and for automatic frequency control of the TCXO generator 302. The CNVR 326 also received the input from the temperature sensor in the radio unit 206. These signals are either received from or transmitted to the ASIC 328.

The ASIC 328 is illustrated in greater detail in Fig. 19. The ASIC 328 includes a transmit (Tx) port interface 332, a receive (Rx) port interface 334, a CNVR interface 336, a Tx control circuit 338, a PLL control 340, a lock indicator 342, a DSP bus interface 344, a Viterbi decoder 346, a timing & interrupt controller 348 and a controller interface 350. The Tx port interface 332 provides digital I and Q signals to the DAC 322. The Rx port interface 334 receives the digital signals from the receiver channels 214 and 216 through the ADC 324. The CNVR interface 336 provides the control signals AGC, AGC-d and AFC to the CNVR 326 and receives the signal OVR from the temperature sensor in the radio unit 206. The Tx control 338 provides control signals for the transmitter 220. The PLL/DDS control 340 provides the control signals to the synthesizer 222 to control the generation of the synthesized frequencies. The DSP bus interface 344 provides an interface between the ASIC 328 and the digital signal processor 330. The Viterbi decoder 346 is utilized to process signals. The timing &

interrupt controller 348 provides timing signals.

Fig. 20 illustrates the BBU controller 228. The controller 228 includes a microcontroller circuit 352, a MMI interface circuit 354, a car accessories interface circuit 356, a microprocessor supervisor 358, a power supply control circuit 360 and a memory circuit 362. The microcontroller 352 is preferably implemented with a Motorola processor MC68302 or an Intel 386EX. The memory 362 is preferably comprised of 128k x 8 static RAM and 512k x 8 flash memory. The memory 362 holds the programs of the microcontroller 352, of the modem 226 and of the VPP 230. The memory 362 will also hold various parameters, user defined telephone numbers and messages that should be kept nonvolatile.

The power supply control circuit 360 monitors the state of the subscriber unit 12 and the state of the car and controls the power supply. The supervisor circuit 358 is responsible for the reset mechanism and the power fail indication. The microcontroller 352 is connected to the MMI interface 354 via a single ON/OFF input and via two RS-232 lines, one for each direction. The RS-232 lines carry data, control and test messages. The car accessories interface circuit 356 provides one input from the car's ignition switch and one output to the car's horn. The microcontroller 352 has a two-wire bi-directional RS-232 connection available for GPS interface.

Referring to Fig. 21, a block diagram of the voice processing package (VPP) 230 is illustrated. The VPP 230 performs encoding and decoding of voice signals and consists of a CODEC 364, an analog interface circuit 366 and a digital signal processor 368. The digital signal processor 368 is preferably an Analog Devices' ADSP2115. The program for the digital signal processor 368 is stored in the memory 362 of the BBU controller 228.

Reference is now made to Figs. 22A and 22B which illustrate a preferred service board found in the subscriber unit.

Figs. 23 and 24 show, respectively, the preferred signal processing performded when transmitting and receiving signals on traffic channels.

Signal Processing

Figs. 25 to 31 illustrate various coding schemes utilized in the present invention. Fig. 25 shows the preferred transmit processing steps for a voice signal when the system is operating in the differential mode. After the signal is voice encoded in step 550, the bits of the signal to be transmitted are divided into classes in accordance with their importance. Those bits with greater importance then receive greater coding. In accordance with a preferred embodiment, four classes (I, II, III and IV) are utilized. The twelve most important bits are assigned to Class I. The next 36 most important bits are assigned to Class II. The next 32 most important bits are assigned to Class III. The least 8 important bits are assigned to Class IV. The bits are then coded with a CRC encoder in step 552 and with a convolutional encoder in step 554 as illustrated. Inband control signals received are also encoded as illustrated. Then, in step 556 the signal is further coded with a partial repetition encoder by the repetition of 10 symbols. In step 558, a symbol is replaced with the value PWR_CNT_SYM with a puncture encoder. The symbol that is replaced is preferably one of the lesser important bits. In step 560 a permutation step can be performed, however, it is presently preferred not to perform this step. In step 562, the interleaving already discussed is performed. In step 564 the frame of data is converted into time slots for transmission.

Fig. 26 illustrates the steps performed by the Rx processing unit in the differential mode. The steps are essentially the opposite of those described with respect to Fig. 25. In step 566, the slot of data is properly framed for processing. In step 568, the data is deinterleaved using the reverse process used during the transmit processing. In step 570, de-permutation is performed if the permutation step was performed during the transmit processing. In step 572, the symbol PWR_CNT_SYM is removed. Then in steps 574 to 576, the signal is decoded as illustrated. In step 578, the bit error rate is calculated and, if too high, can disable the voice reception.

Figs. 27 and 28 illustrate the steps performed by the Tx processing unit and the Rx processing unit, respectively, during the coherent mode of operation. These steps are very similar to those already discussed with respect to Figs. 25 and 26, except the partial repetition encoder repeats a different number of bits during its encoding.

Figs. 29 and 30 illustrate the preferred steps performed when processing data in the differential and in the coherent modes, respectively. Referring to Fig. 29, the data contains a header and one or more frames of data per message. The data is subdivided into a number of frames, each of which is CRC encoded as illustrated. The header is also CRC encoded, as shown. The remaining encoding steps are selectable, the selection depending on the channel conditions. In the low rate mode, the remaining coding includes convolutional encoding (with r = 1/4) and repetition encoding (rate equal 1/2), prior to interleaving, as illustrated. In the medium rate mode, the remaining coding includes convolutional encoding (with r = 1/2) prior to interleaving. In the very high rate mode, the coding rate is one. The coding for the coherent mode is very similar to the coding

utilized in the differential mode, as illustrated in Fig. 30.

The preferred coding scheme utilized for the ACH and the CCH is illustrated in Fig. 31.

Other Processes

Correlations Performed on Received Data

The subscriber unit 12 performs several correlations on the signals it receives in order to achieve proper synchronization with the signal transmitted by the base station 10. In accordance with a preferred embodiment of the present invention, the complex conjugates of the synchronization signal are formed as follows:

H0 = -1 + j; H1 = 1-j; H2 = -1-j; H3 = 1-j; H4 = 1+j; H5 = 1-j; H6 = 1+j; H7 = -1-j; H8 = 1+j; H9 = -1+j; H10 = 1-j; H11 = 1+j; H12 = -1+j; H13 = 1-j; H14 = -1-j; H15 = -1+j; H16 = -1-j;

H17 = -1+j and H18 = -1-j.

The incoming signals are correlated according to the following formulas:

RL_COR0 = The sum of Re(U0(p-j*4))*Re(Hj) - Im(U0(p-j*4))*Im(Hj), where the sum limits are from j = 0 to j = 18;

RL_COR1 = The sum of Re(U1(p-j*4))*Re(Hj) - Im(U1(p-j*4))*Im(Hj), where the sum limits are from j = 0 to j = 18;

where p = 82, the index of the last sample of the synchronization code, U0(n) is the data from the first receiver and U1(n) is the data from the second receiver.

Next, diversity selection is performed, if RL_COR0 is greater than RL-COR1, then RL_COR_PEAK = RL_COR0 and set Ui = U0i for i = 4 to 163, otherwise RL_COR_PEAK = RL_COR1 and set Ui = U1i for i = 4 to 163.

Then the following correlations are performed:

RL_COR_EARLY = RL_COR(p + 1)

RL_COR_LATE = RL_COR(p - 1)

by using Ui with i = 4 to 163.

Note that these correlations are offset in time from the earlier correlations described.

In addition, the following correlation is performed:

IM_COR-PEAK = The sum of Re(U(p-j*4))*Im(Hj)+Im(U(p-j*4))*Re(Hj), where the sum limits are from j = 0 to j = 18.

Preferred methods for performing the above correlations, and for performing channel acquisition, are described in more detail below with reference to Figs. 53 - 55.

Channel Acquisition

One of the processes which the subscriber unit 12 must perform prior to starting communications is the acquisition of the timing of the base station 10. This channel acquisition process is performed when the power to a subscriber unit 12 is first turned on and any time synchronization is lost.

Generally, a subscriber unit 12 performs acquisition by first accessing a table of communication sites which is maintained in the memory of the subscriber unit controller 228. This table contains a list of sites and the frequency of the associated CCHs. The subscriber unit 12 then sequentially scans the frequencies of channels identified by the site table, trying to lock onto each one in turn. In particular, the subscriber unit 12 attempts to lock onto a channel by searching for the synchronization pattern in a SLS slot in the channel. When the subscriber unit 12 scans a frequency, if there is no CCH at that frequency, there will be no SLS to

detect. However, when the subscriber unit 12 scans a control channel (CCH) it will find a SLS. Generally, the subscriber unit 12 performs the scanning by performing a sliding correlation based on the mean square of the phase error between the signals on the channel being scanned and the known sync pattern of the SLS slot which the subscriber unit 12 has stored in its memory. When the subscriber unit 12 finds a high correlation, it has found the SLS slot. The subscriber unit 12 can also search all possible frequencies if the search of known sites is not successful.

Referring to Fig. 32, the process of channel acquisition is illustrated. In step 600, a counter, SLOT_COUNT, is initialized. In step 602, the signals (W1(P)) and the diversity signal W2(P)) being received by the receivers 214 and 216 are first filtered with a digital matched filter and then differentially demodulated. The filter outputs, Z1(p) and Z2(p) and the demodulation outputs U1(p) and U2(p) are output to step 604. In step 604, the average amplitude of these outputs from each receiver channel are compared and the output with the highest average amplitude is selected.

In step 606, the sliding correlation is performed on the data from the selected channel. The correlation measures the mean square phase error between the phase of the received signals and the phase of the synchronization code which is stored in memory.

In step 608, a search is performed for the optimum correlation by looking for the minimum error. The process searches for several minimums in a 216 slot window. Generally, approximately five minimums will be found in this window.

Then, in step 610, the process determines whether the prior steps have been performed the prerequisite number of times. If they have not, then the steps are repeated. If they have, then the process goes

on to step 612. Here, the process verifies that it has locked onto the proper synchronization pattern in the SLS slot. The time index of the minimums in the frame of date has been stored. The time between two minimum values must have a legitimate value (generally 72 slots) to be verified.

In step 614, if it is determined that proper synchronization has not been obtained, the process informs the controller 228 that synchronization has failed. On the other hand, if in step 614, it is determined that proper synchronization has been obtained, then the process continues to step 616. In step 616, the timing of the slot is adjusted in accordance with the indices of the minimum received value. In other words, the synchronization timing is known and the slot timing is determined from this knowledge.

Delay Lock Loop

To lock onto the synchronization information in the SLS slot of the CCH, the subscriber unit 12 performs a delay lock loop process. Referring to Fig. 33, a preferred delay lock loop process for the subscriber unit 12 is illustrated. The correlation values, RL_COR_EARLY and RL_COR_LATE, are utilized in step 618. The following calculation is performed to determine the delay: (RL_COR_EARLY - RL_COR_LATE) / (RL_COR_EARLY + RL_COR_LATE).

When the synchronization information is being correlated, this calculation will yield a result that is nearly zero. On the other hand, if synchronization is not being correlated, then the result will be non-zero.

The result of this calculation is filtered with an infinite impulse response filter in step 620 to provide a smooth response. Then, in step 622, the filtered result is input to a number controlled delay generator where a controlled delay is introduced.

Referring to Fig. 34, a preferred delay lock loop process for the base station 10 is illustrated. Essentially, the same steps 618, 620 and 622 are implemented. However, the base station 10 implements these steps in software so an additional integrating step, step 624, is required.

A preferred implementation of delay lock loop methods is described in more detail below with reference to Figs. 56 - 59.

Automatic Frequency Control

The subscriber unit 12 performs an automatic frequency control process to measure and correct the frequency inaccuracies of its frequency source. Referring to Fig. 35, the steps performed by the subscriber unit 12 during the differential mode are illustrated.

The correlations RL_COR_PEAK and IM_COR_PEAK, of the received data are utilized to determine the errors in the frequency being received by the subscriber unit 12. The error is determined, in step 626, by inputting these values into a table to determine the arctangent of the phase difference between RL_COR_PEAK and IM_COR_PEAK. Then the arctangent is filtered in step 628, as illustrated. The result is utilized to correct the frequency source at the subscriber unit 12.

Initially, it is preferred to estimate and correct the frequency error. This estimation is also utilized as an initial condition at which to start the previously described frequency control process. The first step in the estimation is to select the complex samples of the currently active time slot.

The frequency of the slot and nominal value of the slot frequency are determined and, via software, the several frequency values which vary around the nominal value are substituted into the slot. For example, the

nominal frequency and six frequencies above the nominal may be selected. The mean square phase error of each of these frequencies is calculated and the frequency with the minimum value is utilized.

A preferred implementation of automatic frequency control methods is described in more detail below with reference to Figs. 60 and 61.

Automatic Gain Control

The receivers in the base station 10 and the subscriber unit 12 all need to receive signals within a certain amplitude range in order to be able to optimally perform demodulation. Gain control, therefore, is performed in both the base station 10 and in the subscriber unit 12 to maintain a constant signal level.

Referring to Fig. 36, the gain control process for a receiver channel in the subscriber unit 12 is illustrated. In the first step 630, the average amplitude of each time slot is calculated from the I and Q inputs associated with each symbol. Specifically, this is accomplished by summing the value $r(n) = 1(n) + jQ(n)$ for each time slot n and then dividing by the number of symbols in a time slot, in this case, 39.

Then, in step 632, the average amplitude is compared to a reference level. It is preferred to set the reference level at 0.25, which is 12 dB below the maximum level of 1. The difference between the average amplitude and the reference level is filtered in the next step 634 to smooth the gain control operation to operate on slower variations. The filter is preferably implemented as a standard infinite impulse response filter with a gain of 10 and the coefficients of $C_1 = 0.95$ and $C_2 = 1 \times C_1 = 0.05$. In addition to the filtering, in step 636, it is preferred to limit the range of the signal being filtered to the range of the input signal. In Fig. 36, the signal is limited to the

range of +0.5 to -0.5.

The output of the filter is then sent to a look up table in step 638. The look up table is used to account for the nonlinearities found in the amplifiers in the receivers. Accordingly, the values of the look up table will vary according to the amplifiers being used in the receivers. Then the output of the look up table is converted to an analog signal which is, in turn, applied to the amplifiers in the receivers to effect gain control of the received signal.

In the subscriber unit 12, the steps 630 to 636 are performed in the digital signal processor (DSP, Fig. 15) of the modem 226. The value from the filtering step is output to the receiver channels 214 and 216 (Fig. 16) in the subscriber unit 12 through the ASIC 328 (Fig. 18) and through the CNVR INTERFACE 336 (Fig. 19).

It is preferred to utilize the calculation obtained from a time slot to control the gain of the next slot. Also, the processing is performed separately for each channel, that is, one calculation is performed on the signals received from the receiver channel 214 and another calculation is performed on the signals received from the diversity channel 216.

The previously described steps are performed once the gain control has been performed. During this time, the path to the look up table AGC_TBL is through the limiter 636. During power up, however, the path to the look up table AGC_TBL is through the line labeled SGCS_IC and it is preferred to perform a different gain control process.

Referring to FIG. 37, the preferred steps of the initial gain control process for the subscriber unit 12 are illustrated. These steps set the initial value of SGCS_IC and of the memory Z^{-1} in FIG. 36. In step 640, the initial value of SGCS_IC is set to the middle of the dynamic range of the input signal, in this case 0. Then,

in step 642, the SGCS_IC is applied to the AGC_IC look up table, as illustrated in FIG. 36.

Next, in step 644, a sliding window, preferably of 26 msec. in duration, is established. During a first part of the window, the average value of 164 symbols is determined. Then in the next part of the window, the average value of the next 164 symbols is determined. In a preferred embodiment, this step is repeated a total of 1476 times. The maximum of each of these averages, MAX_W is determined.

Then in step 646, the maximum value from step 644, MAX_W is compared to the predetermined value W1, which is preferably 0.6 and to the predetermined value W2, which is 0.1. If MAX_W is greater than W1, then in step 648, the initial value of SGCS_IC is increased by α. In a preferred embodiment , $\alpha = 0.09$. Then, in step 650, the value of SGCS_IC is checked. If the value is less than or equal to 0.5, then the process returns to step 642. However, if the value is greater than 0.5, then step 652 is performed. In this step, SGCS_IC is set equal to 0.5, the maximum value of the dynamic range of the signal. Then in step 654, the memory Z^{-1} (FIG. 36) is set equal to 0.5 and the signal SGCS_IC is applied to the look up table AGC_TBL.

If in step 646, MAX_W is determined to be less then W0, then in step 646, the initial value of SGCS_IC is decreased by α, which is preferably 0.09. Then, in step 648, the value of SGCS_IC is checked. If the value is greater than or equal to -0.5, then the process returns to step 642. However, if the value is less than -0.5, then step 650 is performed. In this step, SGCS_IC is set equal to -0.5, the minimum value of the dynamic range of the signal. Then in step 654, the memory Z^{-1} (FIG. 36) is set equal to -0.5 and the signal SGCS_IC is applied to the look up table AGC_TBL.

If in step 646, MAX_W is determined to be less

than or equal to 0.6 or greater than or equal to 0.09, then in step 652, the value of SGCS_IC is incremented by a value which is a function of the value of MAX_W. Then in step 654, if SGCS_IC is greater than 0.5, the process goes to step 652. On the other hand, if SGCS_IC is less than or equal to 0.5 in step 654, then the process goes to step 656. In step 656, if SGCS_IC is less than -0.5, the process goes to step 650. On the other hand, if SGCS_IC is greater than or equal to -0.5, then step 654 is performed. In step 654, the memory Z^{-1} (FIG. 36) is set equal to SGCS_IC and the signal SGCS_IC is applied to the look up table AGC_TBL.

The same basic steps are utilized to perform gain control in the base station 10. There are, however, some differences in the implementation details. For example, the averaging step and look up table are implemented in the slot processor while the remaining steps are implemented in the frame processor. Also the initial condition is set via a message on the ACH which set the receiver at the base station 10 at a fixed value.

A preferred implementation of automatic gain control methods is described in more detail below with reference to Figs. 62 - 64C.

<u>Hand OFF</u>

The communication system of the present invention must account for the movement of subscriber units 12 between the sectors 14 to 16. The process by which the movement is accounted for is referred to as the "hand off". It is preferred that the hand off process be seamless, that is, those persons using the system to communicate should not be affected by this process.

Generally, the hand off process includes first detecting when a situation requiring hand off occurs. Simply stated, hand off situations occur in the sectorized system of the present invention when a

subscriber unit 12 relocates from one sector to another, say from sector 14 to sector 15 (FIG. 1) or between any type of site or sector. The next step in the hand off process is to then perform the necessary switching to allow seamless hand off of the communications between the sectors.

In the present invention, the hand off situations are detected by the subscriber unit 12. The subscriber unit 12, by virtue of the channel acquisition procedure and the delay lock loop tracking procedure, is already detecting and locked onto the SLS slots of the CCH in the sector in which it is located, say for example, sector 14. The subscriber unit 12 also searches for the SLS slots of the CCH in the sectors which are adjacent to the sector in which it is located, that is the SLS slots of the CCH in sectors 15 and 16 are searched for. The subscriber unit 12 accomplishes this by referring to the label tag embedded in each SLS and determining the frequencies of the CCHs in the adjacent sectors 15 and 16 of the site 1 or, if the CCHs are all on the same frequencies, by looking at that frequency.

The subscriber unit 12 will, therefore, have three data streams to process -- i.e., the SLS data from the sector in which it is located (sector 14), the data from a first adjacent sector (sector 15) and the data from the second adjacent sector (sector 16). The digital signal processor 226 in the subscriber unit 12 performs an average correlation of synchronization from each of these data streams as compared to the known synchronization pattern to detect when a hand off situation is occurring. The average correlation of the information from the sector 14 is obtained from the correlation processes previously described.

The correlation of the information from the sectors 15 and 16 are somewhat different because the timing of these channels is not known with certainty.

Several correlations are performed on the synchronization information from these sectors based on a prediction of the timing of these synchronizations. Then, for each sector, the correlation with the maximum value is selected. Then, the maximum value for several correlations within that sector are selected and averaged.

While the subscriber unit 12 is in the middle of a sector 14, the average correlation of the synchronization information from that sector will be higher than the average correlation of the synchronization information from the adjacent sectors 15 or 16. However, as the subscriber unit 12 approaches a new sector, say sector 15, the average correlation of the synchronization information from the new sector will be increasing. Eventually, the average correlation of the synchronization information from the new sector 15 will exceed the average correlation of the synchronization information from the old sector 14. When the average correlation of the information from the new sector 15 exceeds the average correlation of information from the old sector 14 by a predetermined amount, the subscriber unit 12 determines that a hand off situation exists.

Once the subscriber unit 12 determines that the hand off situation exists, the controller 228 causes a message to be transmitted on the ACH to the base station 10. The message requests the initiation of a hand off procedure by the base station 10. The base station 10 acknowledges the request by the subscriber unit on the CCH.

Once the base station 10 receives the request of the subscriber unit 12 to initiate a hand off, the base station 10 selects a communication link in the new sector 15 and connects it with the communication line in the old sector 14 via a three way conference bridge found in the PABX 36. In this way, a communication link is

established between the subscriber unit 12 in the old sector 14, the subscriber unit 12 in the new sector 15 and the other person (or machine) communicating with the subscriber unit 12, thereby enabling a seamless hand off.

At this point, the base station 10 and the subscriber unit 12 are both ready to accomplish the hand off. In accordance with the present invention, the hand off is accomplished during a time when there is no activity on the channel. This time is detected by the voice activity detector (VAD) in the vocoder in both that the hand off is done independently in the uplink channel and in the downlink channel. Therefore, the VAD in the base station 10 detects the voice inactivity in the down link communications and, upon detection, causes the hand off to occur between the downlink channels. Also, the VAD in the subscriber unit 12 detects the voice inactivity in the uplink communication and that detection, causes the hand off to occur between the uplink channels.

The actual process by which switching occurs varies. In both in the base station 10 and in the subscriber unit 12, the VAD provides a bit upon detecting voice inactivity to the transmitters. The transmitters, upon sensing the bit from the VAD, switch the transmission from the old sector to the new sector. The receiver hand off is accomplished by transmitting a STOP RECEIVE marker in the TCH which tells the receiver that a talk spurt has ended -- i.e. that there is voice inactivity -- so that it is time to switch to the new sector.

In the event that the receiver misses the marker on the TCH, the hand off can be accomplished by another technique. In accordance with this alternate technique, the communications being received by the receiver includes a CRC code, as previously discussed. This CRC code is analyzed to determine whether the communications has been properly received. For the

hand off process, it is determined how many CRC failures have been detected. When the number of CRC failures exceeds a certain amount during the hand off process (the subscriber unit 12 has requested a hand off), then the hand off is automatically accomplished.

A preferred implementation of hand off methods is described in more detail below with reference to Figs. 65 - 68.

Power Control

The base station 10 and the subscriber unit 12 both perform power control processes whereby each communication link transmits at a minimum power needed for transmission, thereby minimizing the interference in the communication system and saving power. It is preferred to provide power control for the TCH and the ACH, but not for the CCH.

The power control for the downlink TCHs will now be described with reference to FIG. 38. The downlink TCH transmissions (from the base station 10 to the subscriber units 12) start at predetermined power, preferably at the maximum power. In step 660 of the power process, the subscriber unit 12 detects the received power. In a preferred embodiment, the average of the power value (SGCS0 and SGCS1) determined for the two receiver channels 214 and 216 during the automatic gain process are utilized in this process.

Then, in step 662, the average power from step 660 is compared to a predetermined threshold, REF, and the difference is filtered in step 664 so that power is controlled in a smooth fashion. The output of the filter, FIL_SGCS, is then sent to the transmitter in the base station 10 to modify the transmitted power. If there is no activity on the TCH, the information is sent to base station 10 via the ACH. In this case, it is preferred to accumulate the differences between the sensed power and

the predetermined threshold so as to modify the transmitted power in increments, for example by 5 dB. If, however, the TCH is active, then the information is preferably sent via the TCH. In this case, it is preferred that the transmitted information be controlled to modify the transmitted power in small increments, for example by 1 dB.

It is preferred to utilize a different method to control the power transmitted on the ACH due to short message size available on the ACH. The subscriber unit 12 first measures the average received power of the signal being transmitted on the CCH from the base station 10. The CCH transmission power, which is constant, is broadcast and received by the subscriber unit 12. The subscriber unit 12 then subtracts the known CCH transmission power from the average received power to determine the transmission losses on the CCH. The desired reception power at the base station 10 of signals transmitted by the subscriber unit 12 by virtue of a broadcast message. The subscriber unit 12 adds the determined transmission losses on the CCH to the desired reception power to determine the power at which the subscriber unit 12 will transmit on the ACH.

The control of the power transmission on the uplink TCH transmissions (subscriber unit 12 to base station 10) is preferably controlled using a combination of the previously described methods. When the subscriber unit 12 first starts transmitting to the base station 10 on the uplink TCH, the power is controlled in the same way as power is controlled on the ACH. After this starting point, the power on the uplink TCH transmissions is controlled using the same process used to control the power on the downlink TCH transmissions.

A preferred implementation of power control methods is described in more detail below with reference to Figs. 69 - 75.

Reference is now made to Figs. 39 - 48B, which are simplified block diagrams illustrating the structure of the software components of the system of Fig. 1.

Referring again to Figs. 8 - 9, various elements of said Figs. are more fully described as follows:

Elements 56 and 57 of Fig. 9 may be any suitable UHF Antenna, such as model DB 874H105, commercially available from Decibel Products, 3184 Quebec Street, POB 569610, Dallas, TX, USA.

Element 38 of Fig. 8 may be any suitable voice mail system, such as a Trilog system, commercially available from Comverse, Israel.

Element 36 of Fig. 8 may be any suitable PABX system, such as a Coral III or Coral IV system, commercially available from Tadiran, Israel.

Reference is now made to Figs. 49 - 51 which are simplified block diagrams illustrating the structure of a microsite.

Unless otherwise specified, microwave antennas in the apparatus of the present application may be any suitable microwave antenna, such as P/N 8838A-24/PCN, commercially available from Alpha Industries, Inc., Massachusetts, USA.

The apparatus of Figs. 49 - 51 comprises a UHF antenna, which may be any suitable UHF antenna, such as model DB808-Y, commercially available from Decibel Products, 3184 Quebec Street, POB 569610, Dallas, TX, USA.

Conventional techniques for extracting side information (channel state information) include the following publications, all of which are hereby incorporated by reference:

[1] Viljo Hentimen, "A Channel Feedback Communication System", ACTA POLYTECHNICA SCANDINAVICA, Electrical Engineering series No. 26,

Helsinki 1971.
[2] A.J. Viterbi, "A Robust Ratio-Threshold Technique to Mitigate Tone and Partial Band Jamming in Coded MFSK System", MILCOM 82, Boston MA, October 18-20, 1982.
[3] Hyuck M. Kwon, "Imperfect Jamming State Generators for Coded FH/MFSK under Partial-Band Noise Jamming". MILCOM 89, pp. 1.1.1 - 1.1.5.
[4] Hyuck M. Kwon, "Capacity and Cutoff Rate of Coded FH/MFSK Communications with Imperfect Side Information Generators", IEEE Journal on Selected Areas in Comm. Vol. 8, No. 5, June 1990, pp 750-761.
[5] Hyuck M. Kwon, Pil Joong Lee, "Combined Tone and Noise Jamming Against Coded FH/MFSK ECCM Radios", IEEE Journal on Selected Areas in Comm. Vol. 8, No. 5, June 1990, pp. 871-883.
[6] R.R. McKerracher, P.H. Wittke, "Frequency Hopped Spread Spectrum Systems with Reed Solomon Coding and Practical Jammer State Estimation", MILCOM 92, pp. 16.6.1 - 16.6.6.
[7] G.L. Stuber, Jon W. Mark, Ian F. Blake, "Diversity and Coding for FH/MFSK Systems with Fading and Jamming - Part II: Selection Diversity", IEEE Trans. on Comm., Vol. 37, Aug. 89, pp 859-869.[8] Joseph M. Hanraty, Gordon L. Stuber, "Performance Analysis of Hybrid ARQ Protocols in a Slotted Direct Sequence Code-Division Multiple Access Network Jamming Analysis", IEEE Journal on Selected Areas in Comm., Vol.8, No. 4, May 1990.

Reference is now made to Fig. 52, which is a simplified block diagram illustrating the structure of a remote sector. A remote sector, also termed herein a "remote station", functions to enhance the coverage of the system.

Time Alignment Method

In accordance with a preferred embodiment of the present invention, the system is operative to

straighten out the timing of all of the signals received by the base station.

Generally, subscribers are at different distances. They all downlink and are synchronized. They send uplink on the synchronized timing but due to different distances the communications are not received in synchronized timing.

A preferred method to make sure that they are received with synchronize timing is as follows: The base receiver measures the time of arrival on the uplink, and checks if they are within a predefined window. The base station tells the remotes to change their transmission timing so as to have the received signals within the window of the synchronized timing of the base station.

A preferred implementation of time alignment methods is described in more detail below with reference to Figs. 76 and 77.

MARQ Method

In accordance with a preferred embodiment of the present invention, the system is operative to reduce the need for large number of multiple retransmissions when long digital messages are being sent.

A preferred method for reducing the large number of retransmissions is as follows: Break received message into segments and flag only those segments which are not received correctly. Upon each subsequent transmission only replace those flagged portions; once all flagged portions are received correctly, there is no need for further retransmission even though the retransmission may not have been received totally completely.

A preferred implementation of Marq methods is described in more detail below with reference to Figs. 78 - 81.

Methods for coordinated hits and forced erasures are discussed below with reference to Figs. 82 - 86.

Methods for talk around are discussed below with reference to Figs. 87 - 92.

Methods for service at the foot of a base station are discussed below with reference to Figs. 93 - 97.

Reference is now made to Fig. 53 which is a generalized block diagram illustration of a portion of a subscriber unit in a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention.

A frequency hopping multiple access communication system, in accordance with a preferred embodiment of the present invention, utilizes a frequency hopping multiple access communication network and a multiplicity of base stations, at least some of which receive and transmit information at a plurality of radio frequencies over the frequency hopping multiple access communication network.

The system also includes a multiplicity of subscriber units, each receiving and transmitting information at a plurality of radio frequencies via the frequency hopping multiple access communication network.

At a subscriber unit, a receiving and transmitting unit, generally indicated by reference numeral 1010, is operable to receive and transmit radio-frequency (RF) signals.

Receiving and transmitting unit 1010 preferably includes a first antenna 1012, a second antenna 1014 and a radio unit (RU) 1016. Antennas 1012 and 1014 are operable to establish communication channels with a base station (not shown). Preferably, antennas 1012 and 1014 operate in the frequency range 890 - 950 MHz.

In radio unit 1016, RF signals are received at receivers 1018 and 1020, also referred to as RXD and RX respectively. Receiver 1018 is coupled to antenna 1012 and receiver 1020 is coupled to antenna 1014 via a duplexer unit 1022. Preferably, receivers 1018 and 1020 are RF/IF converters which convert RF signals to intermediate frequency (IF) signals.

In a transmission mode, only antenna 1014 is employed. In a reception mode however, both antenna 1012 and 1014 are employed to achieve space diversity. In that case, receivers RXD and RX determine the quality of reception of the corresponding received signals and the best of the corresponding received signals are selected for processing.

Preferably, receivers 1018 and 1020 are coupled to a combined gain and frequency control unit 1024 which is operable to provide separate automatic gain control (AGC) signals and common automatic frequency control (AFC) signals to receivers 1018 and 1020. In an alternative embodiment of the present invention gain and frequency control unit 1024 is not a combined unit but may rather include a separate gain control unit and a separate frequency control unit.

Gain and frequency control unit 1024 is coupled to a synthesizer unit 1026 which is coupled to a transmitter 1028. Transmitter 1028 is coupled to a power control unit 1030. Gain and frequency control unit 1024 provides signals to synthesizer unit 1026 and receives inputs, including a clock signal and data, from a modem 1034 in a baseband unit (BBU) 1032.

Receivers 1018 and 1020 are also coupled to synthesizer unit 1026 which generates signals necessary to downconvert signals received by receivers 1018 and 1020. Receivers 1018 and 1020 provide the downconverted signals of an intermediate frequency (IF) to modem 1034 in BBU 1032.

Synthesizer unit 1026 is also operable to generate signals necessary to modulate signals transmitted by transmitter 1028. Synthesizer unit 1026 also includes a local oscillator 1036 which is operable to supply a signal of a given frequency to the RF/IF converters (receivers 1018 and 1020) in response to a control signal received from modem 1034.

The signals generated by synthesizer unit 1026 to frequency downconvert the signals received by receivers 1018 and 1020, and the signals generated by synthesizer unit 1026 to frequency upconvert the signals transmitted by transmitter 1028 are preferably hopping signals which are generated in accordance with control signals communicated to and from modem 1034 in BBU 1032. Different hopping signals may be generated for transmission and for reception.

Transmitter 1028 receives control signals from modem 1034 in BBU 1032 and power control signals from power control unit 1030, which also receives inputs from modem 1034 in BBU 1032. Transmitter 1028 is also coupled to a service board (not shown) which provides electric power to transmitter 1028. Transmitter 1028 outputs data for transmission to antenna 1014 via duplexer unit 1022.

Preferably, modem 1034 also includes a DSP (digital signal processing) unit 1038 and an ASIC (Application Specific Integrated Circuit) unit 1040 which is coupled to DSP unit 1038. Alternatively, DSP unit 1038 and ASIC unit 1040 may be separate units which are coupled to modem 1034. It is to be appreciated that DSP unit 1038 is operable to control the operation of ASIC unit 1040.

In a preferred embodiment of the present invention an initialization algorithm for acquiring a precise frequency and a precise timing of a control channel is performed in DSP unit 1038 as described herein after. The initialization algorithm is utilized, as

described herein after, in conjunction with an FFT (fast Fourier transform) computing module 1042 which may form part of DSP unit 1038 or may be a separate module coupled to DSP unit 1038.

Reference is now made to Fig. 54 which is a flow chart illustration describing the operation of frequency acquisition in a channel feature acquisition algorithm that is performed at the subscriber unit of Fig. 53.

In a preferred embodiment of the present invention when communication is initiated, the subscriber unit of Fig. 53 receives RF signals via antennas 1012 and 1014. The signals are received at receivers 1018 and 1020 and provided to modem 1034. DSP unit 1038 in modem 1034 is operative to generate an estimated frequency offset value for a signal received by modem 1034.

The estimated frequency offset value is provided to synthesizer 1026 which generates a frequency converter control signal that changes the frequency of local oscillator 1036. The frequency converter control signal is provided to receivers 1018 and 1020 and is operative to cancel the estimated frequency offset.

Typically, the signals which are received over the control channel and provided to modem 1034 are modulated signals. Preferably, modem 1034 is operable to remove the modulation of the signal, to thereby generate a carrier wave. It is to be appreciated that the carrier wave may be a distorted carrier wave.

FFT computing module 1042 in DSP unit 1038 is operable to convert the waveform of the carrier wave from a time domain to a frequency domain thus generating a final spectrum function in the frequency domain. FFT computing module 1042 provides the final spectrum function to DSP unit 1038 which computes a frequency offset by finding a frequency value that maximizes the final spectrum function.

The conversion of the waveform of the carrier wave from a time domain to a frequency domain is done portion by portion for a plurality of portions of the waveform. Such conversion generates a plurality of intermediate spectrum functions.

The intermediate spectrum functions are combined in a spectrum function combining module which forms part of FFT computing module 1042. The intermediate spectrum function values are summed in a summing module for each of a multiplicity of frequencies to form the final spectrum function.

Reference is now made to Fig. 55 which is a flow chart illustration describing timing acquisition in a channel feature acquisition algorithm which is performed at the subscriber unit of Fig. 53.

In a preferred embodiment of the present invention receivers 1018 and 1020 of Fig. 53 provide signals to DSP unit 1038 in modem 1034. At DSP unit 1038 a synchronization code embedded in the received signal is detected and timing information associated with the detected synchronization code is generated. It is to be appreciated that the received signal may include a synchronization code period and at least one synchronization code is embedded in the synchronization code period.

The timing information is provided to ASIC unit 1040 which includes a local timing system and a local timing system synchronizer which synchronize the local timing system in accordance with the timing information.

Preferably, a representation of the synchronization code is stored in a memory (not shown) at DSP unit 1038. As described herein after, the representation of the synchronization code in the memory is correlated with the received signal.

In accordance with a preferred embodiment of the invention time windows are set and a sliding

correlation operation with the stored representation of the synchronization code is performed within a time window in which at least one element of the synchronization code is known to appear. A correlation search over the entire window is performed, and any optimal correlation, if found, is stored in the memory. It is to be appreciated that the timing information associated with the synchronization code includes an indication of a time at which the optimal correlation appears.

The sliding correlation operation may be performed within a window in which at least N elements of the synchronization code are known to appear and M optimal correlations within that window may be found. In such a case the timing information may include an indication of a time at which a predetermined one from among the M optimal correlations appears.

The M optimal correlations may correspond to K <= M points in time. The K optimal correlations are then verified by comparing the K points in time to a known timing pattern of the K optimal correlations. The K optimal correlations may form a subset of the M optimal correlations.

If the correlations are not verified a fail signal is generated. If the correlations are verified a slot timing adjustment is performed in accordance with the optimal correlations found.

It is to be appreciated that adjustment of timing is also required for a timing frame which is larger than a slot. Such a timing frame is indicated as a super-frame, and it may include a frame of 4.8 seconds.

Typically, a super-frame includes a plurality of synchronization code periods, such as 10 synchronization code periods. At least one synchronization code is embedded within each of the plurality of synchronization code periods, and each

synchronization code is associated with a synchronization code label which includes information indicating the location of the synchronization code in the series of synchronization codes forming part of the super frame.

In a preferred embodiment of the invention each label includes a constant portion and a variable portion, wherein the variable portion includes a sequence of labels which determine the location of each slot associated with a label in the super frame. To verify the labels, a time window which may include several slots is set and the existence of constant portions is checked. Additionally, correspondence of the sequence of labels to a preselected monotonically ordered sequence of numbers is searched.

If correspondence of at least a portion L of the labels in the sequence of labels with a portion of the preselected monotonically ordered sequence of numbers is found, and the constant portions in the L labels are identical, then the labels are verified and a timing adjustment of the super-frame is performed by changing, in software, a slot counter. If verification fails, a fail signal is generated.

Reference is now made to Fig. 56 which is a generalized block diagram illustration of a portion of a subscriber unit in a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention.

The frequency hopping multiple access communication system preferably utilizes a frequency hopping multiple access communication network and a multiplicity of base stations, at least some of which receive and transmit information at a plurality of radio frequencies over the frequency hopping multiple access communication network.

The system also includes a multiplicity of

subscriber units, each receiving and transmitting information at a plurality of radio frequencies via the frequency hopping multiple access communication network.

At a subscriber unit, a receiving and transmitting unit, generally indicated by reference numeral 2010, is operable to receive and transmit radio-frequency (RF) signals.

Receiving and transmitting unit 2010 preferably includes a first antenna 2012, a second antenna 2014 and a radio unit (RU) 2016. Antennas 2012 and 2014 are operable to establish communication channels with a base station (not shown). Preferably, antennas 2012 and 2014 operate in the frequency range 890 - 950 MHz. However, tuning to other frequency ranges is also possible.

In radio unit 2016, RF signals are received at receivers 2018 and 2020, also referred to as RXD and RX respectively. Receiver 2018 is coupled to antenna 2012 and receiver 2020 is coupled to antenna 2014 via a duplexer unit 2022. Preferably, receivers 2018 and 2020 are converters which convert RF signals to intermediate frequency (IF) signals.

In a transmission mode, only antenna 2014 is employed. In a reception mode however, both antenna 2012 and 2014 are employed to achieve space diversity. In that case, receivers RXD and RX determine the quality of reception of the corresponding received signals and the best of the corresponding received signals are selected for processing.

Receivers 2018 and 2020 are coupled to a combined gain and frequency control unit 2024 which is operable to provide separate automatic gain control (AGC) signals and common automatic frequency control (AFC) signals to receivers 2018 and 2020. In an alternative embodiment of the present invention gain and frequency control unit 2024 is not a combined unit but may rather include a separate gain control unit and a separate

frequency control unit.

Gain and frequency control unit 2024 also provides signals to a synthesizer unit 2026, and receives inputs, including a clock signal and data, from a modem 2030 in a baseband unit (BBU) 2028.

Receivers 2018 and 2020 are also coupled to synthesizer unit 2026 which generates signals necessary to downconvert signals received by receivers 2018 and 2020. Receivers 2018 and 2020 provide the downconverted signals to modem 2030 in BBU 2028.

Synthesizer unit 2026 is also operable to generate signals necessary to modulate signals transmitted by a transmitter 2032 which also forms part of radio unit 2016.

The signals generated by synthesizer unit 2026 to downconvert the signals received by receivers 2018 and 2020, and the signals generated by synthesizer unit 2026 to modulate the signals transmitted by transmitter 2032, are preferably hopping signals which are generated in accordance with control signals communicated to and from modem 2030 in BBU 2028. Different hopping signals are generated for transmission and for reception.

Transmitter 2032 receives control signals from modem 2030 in BBU 2028 and gain control signals from a gain control unit 2024, which also receives inputs from modem 2030. Transmitter 2032 is also coupled to a service board (not shown) which provides electric power to transmitter 2032. Transmitter 2032 outputs data for transmission to antenna 2014 via duplexer unit 2022.

It is to be appreciated that in a frequency hopping system a timing synchronization signal transmitted by a base station does not precisely match a local timing synchronization signal generated in the subscriber unit. A delay locked loop, as described with reference to Fig. 57, is operable to maintain the synchronization between the timing signal transmitted by

the base station and the timing signal generated in the subscriber unit.

Preferably, the software portion of the delay locked loop is performed and utilized in a DSP (Digital Signal Processing) unit 2036, which forms part of modem 2030, and the hardware operations of the delay locked loop are performed and utilized in a timing unit (not shown) which is part of an ASIC (Application Specific Integrated Circuit) board 2038 in modem 2030 of BBU 2028. In a preferred embodiment of the present invention DSP unit 2036 and ASIC board 2038 may be separate units which are coupled to modem 2030.

Reference is now made to Fig. 57 which is a simplified illustration of the operation of a delay locked loop in a subscriber unit which forms part of a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention.

A subscriber unit performs several correlations on a received signal in order to achieve proper synchronization with a signal transmitted by a base station over a control channel. In accordance with a preferred embodiment of the present invention, an ideal waveform of a synchronization code signal is stored at DSP unit 2036 of Fig. 56. As described hereinafter, a synchronization signal portion of incoming RF signals that are received at receivers 2018 and 2020 is detected at the output of receivers 2018 and 2020, and correlated at DSP unit 2036 with the stored ideal waveform of the synchronization code signal.

At the subscriber unit, a synchronization/label slot monitoring task is performed which detects, in a synchronization/label slot (SLS) in the control channel, a timing synchronization signal which includes a pattern of symbols having a predetermined structure. Additionally, the SLS monitoring task sets a correlation

flag COR_FLAG0, where COR_FLAG0=0 indicates an invalid correlation result and COR_FLAG0=1 indicates a valid correlation. It is to be appreciated that the delay locked loop algorithm is performed only when COR_FLAG0=1.

The timing synchronization signal received at the SLS is correlated with the ideal waveform of the synchronization code signal, which is stored at the subscriber unit, to provide an early correlation value RL_COR_ERLY and a late correlation value RL_COR_LATE respectively.

The early and late correlation values are a result of the correlations performed at the times t1 and t2, where t1 is a time preceding the estimated time at which the synchronization code embedded in the output signal of the RF/IF converter is maximally correlated with the ideal waveform of the synchronization code signal and t2 is a time following that estimated time.

The correlation values RL_COR_ERLY and RL_COR_LATE are fed to a delay detector 2050 in which a normalized difference between the early correlation value and the late correlation value is generated. As described hereinafter, the normalized difference is employed to generate a control signal which is monotonically related to the difference.

The normalized difference value is found by employing the following equation:

D_ERR = (RL_COR_ERLY - RL_COR_LATE) / (ABS(RL_COR_ERLY) +
ABS(RL_COR_LATE)) (1)

where ABS is an absolute value.

The output of delay detector 2050 is a time delay value D_ERR that is provided to a loop filter 2052 which is an infinite impulse response filter of a lead-leg type.

Loop filter 2052 receives input values including constant coefficient values C1, C2 and C3 and an initial condition value IC=0 for integration. Preferably, C1=1.526, C2=0.0968 and C3=1.

In loop filter 2052, D_ERR is multiplied by C2 and summed with an iterated feedback value DLL_INTEG which is integrated with an initial condition IC=0 and multiplied by C3. The sum is added to the value of the multiplication of D_ERR by C1. The output of loop filter 2052 is a smooth response signal DLL_FIL. The following equations are employed to compute DLL_FIL:

$$DLL\_INTEG = D\_ERR * C2 + DLL\_INTEG * C3 \quad (2)$$
$$DLL\_FIL = D\_ERR * C1 + DLL\_INTEG \quad (3)$$

Smooth response signal DLL_FIL is provided to a number controlled delay generator 2054 which is operable to generate a hardware delay control signal (HDCS) that is employed to correct timing synchronization.

In number controlled delay generator 2054 the smooth response signal DLL_FIL is summed with a feedback value which is integrated with an initial condition IC=0 and summed with a rounded value of a previous iterated feedback value NCD. The rounded value of the previous iterated feedback value is achieved by dividing NCD by a coefficient R, rounding-off the result and multiplying HCDS by the coefficient R, where R=1.

The output of the loop is the hardware delay control signal (HDCS) which is monotonically related to the difference between the timing of the IF signal generated by the receivers 2018 and 2020 and the timing of the ideal waveform of the synchronization signal of the local (subscriber) timing system. HDCS is employed as a hardware instruction to determine the symbol timing by controlling the size of the dividers in counters used for

such determination.

The following equations are employed to compute HDCS:

NCD = DLL_FIL + X (4)

HDCS = ROUND (NCD/R) (5)

X = NCD + R * HDCS (6)

In a preferred embodiment of the present invention HDCS is an integer number. If HDCS is not equal to zero then the timing signals generated in a receiving mode and the timing signals generated in a transmitting mode are shifted by HDCS time increments in the appropriate direction in accordance with the sign of HDCS. If shifts smaller than a time increment are required (HDCS=0), then such shifts are accumulated until at least one full time increment shift may be applied. Preferably, the time increments at the subscriber unit are substantially equal to 3 microseconds.

It is to be appreciated that all previously mentioned coefficients and correlation values employed in the delay locked loop are real numbers, except for HDCS which is an integer.

Reference is now made to Fig. 58 which is a generalized block diagram illustration of a portion of a base station 2100 in a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention.

The base station includes a group of receivers 2102, each coupled to an antenna 2104. Preferably, each receiver in the group of receivers 2102 is operable to receive information signals over one frequency channel.

The group of receivers 2102 is coupled to a group of slot processors (SP) 2106. Each receiver in the group of receivers 2102 is coupled to a separate slot processor in the group of SP 2106.

The group of SP 2106 is coupled to a

communication bus 2108, preferably a modified HDLC communication bus.

A group of frame processors (FP) 2110 is also coupled to communication bus 2108. The frame processors in the group 2110 receive inputs from the slot processors in the group 2106 via communication bus 2108.

Reference is now made to Fig. 59 which is a simplified illustration of the operation of a delay locked loop in a base station of a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention.

The delay locked loop algorithm performed at the base station 2100 is similar to the delay locked loop algorithm performed at the subscriber unit 2010, except that the timing synchronization signal is received and detected at a UTCH_SYNC slot and the calculation of D_ERR is performed in a slot processor and fed as an input to the loop. Furthermore, an additional integration step is applied in the loop due to implementation of the control of the delay in software at the base station.

D_ERR is fed to a loop filter 2140 which is similar to loop filter 2052, as mentioned previously with reference to Fig. 57, except for the coefficients values C1 and C2 which, at the base station, have the values C1=0.44 and C2=0.0195. The coefficient C3 has the same value as that of the subscriber unit. The output of loop filter 2140 is a smooth response signal DLL_FIL which is fed to a number controlled delay generator 2142.

Number controlled delay generator 2142 is identical with number controlled delay generator 2054 as mentioned previously with reference to Fig. 57. The output of number controlled delay generator 2142 is a DL_OUT value that is fed to an additional integration loop 2144 which is operable to provide a software delay control signal (SDCS), and a hardware delay control

signal (HDCS). The value of SDCS is limited between the values +/- SDCS_LIM, where SDCS_LIM is preferably set to 21.

The additional integration loop 2144 is employed at the base station to convert frequency shifts to time shifts. HDCS which is the output of additional integration loop 2144, is employed to correct timing synchronization in software by controlling shift of complex samples in a slot or in a buffer (not shown) in accordance with the time shifts calculated at the additional integration loop 2144.

Additional integration loop 2144 receives as input an initial condition value IC, where IC=BACH_DLY and BACH_DLY is a value received from an algorithm which calculates an initial time of the loop and is performed in a frame processor which processes a control channel or a frame processor which processes an access channel.

In a preferred embodiment of the present invention HDCS is an integer number. If HDCS is not equal to zero then the timing signals generated in a receiving mode and the timing signals generated in a transmitting mode are shifted by HDCS time increments in the appropriate direction in accordance with the sign of HDCS. If shifts smaller than a time increment are required (HDCS=0), then such shifts are accumulated until at least one full time increment shift may be applied. Preferably, the time increments at the base station are substantially equal to 6 microseconds.

The previously mentioned coefficients and correlation values employed in the delay locked loop are real numbers, except for HDCS, SDCS and BACH_DLY which are integers.

Reference is now made to Fig. 60 which is a generalized block diagram illustration of a portion of a subscriber unit in a frequency hopping multiple access communication system constructed and operative in

accordance with a preferred embodiment of the present invention.

A frequency hopping multiple access communication system, in accordance with a preferred embodiment of the present invention, utilizes a frequency hopping multiple access communication network and a multiplicity of base stations, at least some of which receive and transmit information at a plurality of radio frequencies over the frequency hopping multiple access communication network.

The system also includes a multiplicity of subscriber units, each receiving and transmitting information at a plurality of radio frequencies via the frequency hopping multiple access communication network.

At a subscriber unit, a receiving and transmitting unit, generally indicated by reference numeral 3010, is operable to receive and transmit radio-frequency (RF) signals.

Receiving and transmitting unit 3010 preferably includes a first antenna 3012, a second antenna 3014 and a radio unit (RU) 3016. Antennas 3012 and 3014 are operable to establish communication channels with a base station (not shown). Preferably, antennas 3012 and 3014 operate in the frequency range 890 - 950 MHz. However, tuning to other frequency ranges is also possible.

In radio unit 3016, RF signals are received at receivers 3018 and 3020, also referred to as RXD and RX respectively. Receiver 3018 is coupled to antenna 3012 and receiver 3020 is coupled to antenna 3014 via a duplexer unit 3022. Preferably, receivers 3018 and 3020 are converters which convert RF signals to intermediate frequency (IF) signals.

In a transmission mode, only antenna 3014 is employed. In a reception mode however, both antenna 3012 and 3014 are employed to achieve space diversity. In that case, receivers RXD and RX determine the quality of

reception of the corresponding received signals and the best of the corresponding received signals are selected for processing.

Receivers 3018 and 3020 are coupled to a combined gain and frequency control unit 3024 which is operable to provide separate automatic gain control (AGC) signals and common automatic frequency control (AFC) signals to receivers 3018 and 3020. In an alternative embodiment of the present invention gain and frequency control unit 3024 is not a combined unit but may rather include a separate gain control unit and a separate frequency control unit.

Gain and frequency control unit 3024 is coupled to a synthesizer unit 3026 which is coupled to a transmitter 3028. Transmitter 3028 is coupled to a power control unit 3030. Gain and frequency control unit 3024 provides signals to synthesizer unit 3026 and receives inputs, including a clock signal and data, from a modem 3034 in a baseband unit (BBU) 3032.

Receivers 3018 and 3020 are also coupled to synthesizer unit 3026 which generates signals necessary to downconvert signals received by receivers 3018 and 3020. Receivers 3018 and 3020 provide the downconverted signals of the intermediate frequency (IF) to modem 3034 in BBU 3032.

Synthesizer unit 3026 is also operable to generate signals necessary to modulate signals transmitted by transmitter 3028.

The signals generated by synthesizer unit 3026 to downconvert the signals received by receivers 3018 and 3020, and the signals generated by synthesizer unit 3026 to modulate the signals transmitted by transmitter 3028 are preferably hopping signals which are generated in accordance with control signals communicated to and from modem 3034 in BBU 3032. Different hopping signals may be generated for transmission and for reception.

Transmitter 3028 receives control signals from modem 3034 in BBU 3032 and power control signals from power control unit 3030, which also receives inputs from modem 3034 in BBU 3032. Transmitter 3028 is also coupled to a service board (not shown) which provides electric power to transmitter 3028. Transmitter 3028 outputs data for transmission to antenna 3014 via duplexer unit 3022.

It is to be appreciated that in a frequency hopping system the frequencies of the received signal may be slightly different from the nominal transmitted frequencies due to errors resulting from finite RF source accuracies at the subscriber unit and Doppler shifts. The gain and frequency control unit 3024 is operable to determine and reduce inaccuracies in each separate frequency of a hopping signal received from at least one base station to acceptable values by employing an AFC algorithm as described with reference to Fig. 61.

The AFC algorithm is performed in a DSP (digital signal processing) unit 3036 which may be part of modem 3034. Alternatively, DSP unit 3036 may be a separate unit which is coupled to modem 3034.

Reference is now made to Fig. 61 which is a simplified illustration of the operation of automatic frequency control in a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention.

A subscriber unit performs several correlations on a received signal in order to achieve proper synchronization with a signal transmitted by a base station. In accordance with a preferred embodiment of the present invention, an ideal waveform of a synchronization code signal is stored at DSP unit 3036 of Fig. 60. A synchronization signal portion of incoming RF signals which are received at receivers 3018 and 3020 and are detected and correlated at DSP unit 3036 with the stored

ideal waveform of the synchronization code signal to provide complex correlation signals.

The correlation provides a frequency difference which is employed to generate a control signal, monotonically related to the frequency difference. The frequency offset is also employed to control the operation of a local frequency source, typically a local oscillator (LO) (not shown) in synthesizer unit 3026, and to compensate for a nonlinearity of operation of the local frequency source.

As mentioned before, with reference to Fig. 60, in a reception mode both antennas of the subscriber unit are employed to achieve space diversity, and the best received signal is selected for processing. Thus, correlation signals are obtained at both antennas of the subscriber unit, and diversity selection is performed in DSP unit 3036 on the correlation signals from both antennas to provide a best complex peak correlation signal COR_PEAK, also indicated by (RL_COR_PEAK, IM_COR_PEAK).

Additionally, the diversity selection sets a correlation flag COR_FLAG0 where COR_FLAG0=0 indicates an invalid correlation result and COR_FLAG0=1 indicates a valid correlation. It is to be appreciated that automatic frequency control is performed only when COR_FLAG0=1.

The frequency offset between the signal received from the base station and the frequency of the LO in the subscriber unit may be obtained by determining the angle between the complex number COR_PEAK and a positive real axis. The angle obtained is employed to determine an error correction voltage which is applied to a voltage controlled oscillator (not shown), in order to reduce the frequency error.

The imaginary part of the complex peak correlation signal IM_COR_PEAK is provided to a first AFC filter 3100 which is performed in DSP unit 3036.

In first AFC filter 3100 the IM_COR_PEAK signal is filtered to minimize the effects of noise and interference. Filtering of the signal is achieved by performing iterations including integration over time and feed-back operations.

For initial conditions in these operations the coefficients C1, C2 and INTEG1_IC are employed, with INTEG1_IC=0 prior to the first iteration. It is to be appreciated that C1 and C2 are real numbers whose values are determined in accordance with the number of iterations performed as set forth in Table 1 which is described herein below.

The output of the first AFC filter 3100 is a preliminary filtered signal (real number) indicated by FIL_OUT. Signal FIL_OUT is multiplied by a real number coefficient C3 whose values are determined in accordance with the number of iterations performed as set forth in Table 1 which follows:

TABLE 1

NO. OF ITERATIONS	C3	C2	C1
1...20	1	364/32768	0.2502
21...40	1/2	182/32768	0.2502
41...60	1/3	121/32768	0.2502
61...80	1/4	91/32768	0.2502
81...100	1/5	73/32768	0.2502
101...120	1/6	61/32768	0.2502
121...140	1/7	52/32768	0.2502
141...160	1/8	45/32768	0.2502
161...180	1/9	40/32768	0.2502
181...Forward	1/10	36/32768	0.2502

The output after the multiplication with C3 is provided to a second AFC filter 3102 which is performed in DSP unit 3036. Second AFC filter 3102 receives an

input signal SFCS_IC (real number) from DSP unit 3036 which is kept limited between 0 and 1 by a limiter 3104. AFC filter 3102 is operable to perform additional iterations including integration over time using an intermediate value INTEG2, and feed-back using the SFCS_IC signal.

The output of the second AFC filter 3102 is a smoothed frequency offset signal, indicated by SFCS (Software Frequency Control Signal) which has a real number value. The SFCS signal is provided to a frequency to voltage converter 3106, also referred to as CONV, in gain and frequency control unit 3024.

The SFCS signal is employed to determine an error correction voltage signal HFCS (Hardware Frequency Control Signal) which is applied to a voltage controlled oscillator (not shown) in gain and frequency control unit 3024, via a D/A converter (not shown) in gain and frequency control unit 3024, in order to reduce the frequency error of the LO. The HFCS signal is an integer number which is obtained by truncating the function SFCS * (2**12 - 1).

In a case of an overflow in the algorithm, variables which exceed +1 or -1 are set to +1 and -1 respectively.

Reference is now made to Fig. 62 which illustrates gain control apparatus which is constructed and operative in accordance with a preferred embodiment of the present invention. The gain control apparatus of Fig. 62 is generally operative to vary the degree of amplification of an amplifier in a radio communications system. Specifically, the apparatus of Fig. 62 is intended for use in a slotted radio communication system, where received signals are divided into slots. Examples of slotted radio communication systems include time-division multiple-access systems. Examples also include certain frequency-division multiple-access systems and

cellular communication systems, when those systems also are slotted.

The apparatus of Fig. 62 comprises a sample processor 4100. The sample processor 4100 may be any suitable sample processor as, for example, a suitably programmed digital signal processor (DSP), programmed to compute the average of the absolute value of the complex value of incoming samples of each slot.

The sample processor 4100 receives gain controlled samples representing the gain level, at different times, of a radio signal received by a radio receiver within a slotted radio communication system. The gain controlled signals received by the sample processor 4100 may be generated, for example, by the RF/IF stage of a radio and an analog to digital converter.

Preferably, the gain controlled samples comprise a plurality of samples for a current slot as, for example, 39 samples per slot. In one preferred embodiment of the present invention, when the received signal comprises a plurality of symbols, there is one sample for each of the plurality of symbols.

As explained below with reference to Fig. 63, the sample processor 4100 processes the gain controlled samples to produce a processed signal.

The apparatus of Fig. 62 further comprises error determination apparatus 4110. The error determination apparatus 4110 may be any suitable error determination apparatus as, for example, a suitably programmed DSP.

The error determination apparatus 4110 receives the processed signal from the sample processor 4100. As explained below with reference to Fig. 63, the error determination apparatus 4110 determines the error in the processed signal by comparing the level of the processed signal to a reference level. The error determination

apparatus 4110 produces an error signal representing the desired attenuation of the received signal in dB units.

The apparatus of Fig. 62 further comprises control apparatus 4120. The control apparatus 4120 may be any suitable control apparatus as, for example, a suitably programmed DSP.

The control apparatus 4120 receives the error signal from the error determination apparatus 4110. The control apparatus 4120 converts the error signal, representing attenuation, to a gain control signal and then, through a digital to analog converter, produces an electrical signal of a voltage representing the desired attenuation.

The apparatus of Fig. 62 further comprises an attenuator 4130. The attenuator 4130 may be any suitable voltage controlled attenuator, such as, for example, a model TQ9114N commercially available from Triquint Semiconductor, Wireless Communication Division, 3625A W. Murray Blvd, Beaverton, OR 97005. The attenuator 4130 receives the analog electrical signal from the control apparatus 4120 and attenuates the received signal accordingly.

The operation of the apparatus of Fig. 62 is now briefly described. Reference is now additionally made to Fig. 63, which is a simplified flowchart illustrating the operation of a portion of the apparatus of Fig. 62. The flowchart of Fig. 63 illustrates the operation of the sample processor 4100 and the error determination apparatus 4110 of Fig. 62. The method of Fig. 63 preferably comprises the following steps:

Operation of the sample processor 4100:

STEP 4150: Average computation

The plurality of gain controlled samples received by the sample processor 4100, as described above with reference to Fig. 62, are averaged together, typically using an arithmetic average. Preferably, the

samples are grouped together into groups, each group comprising samples for a particular slot. Typically, the arithmetic average is computed as the average of the absolute value of the amplitude level over all samples in an active slot. Typically, there are a fixed number of sample per active slot, such as 39 samples per active slot.

Operation of the error determination apparatus 4110:

STEP 4160: Logarithmic scaling

The averaged output of step 4150 is scaled logarithmically to decrease sensitivity to variations. A typical logarithmic scaling would be of the form a log(x) + b , where: x is the averaged output of step 4150; a is a dB scaling factor, typically equal to 20; and b is a threshold level representing the desired level, typically equal to 12.

STEP 4170: Filtering

The logarithmically scaled output of step 4170 is then filtered, preferably with a finite impulse response - FIR filter. The purpose of step 4170 is to smooth the output, reducing or eliminating the effects of noise and interference.

An example of a detailed implementation of automatic gain control method for the apparatus of Fig. 62, comprising the method of Fig. 63, is described in Appendix A.

Reference is now made to Fig. 64A, which is a simplified flowchart illustrating an initialization method for the apparatus of Fig. 62, as described in detail in Appendix B. Specifically, steps 4190, 4200, 4210, 4220, 4230, 4240, and 4250 of Fig. 64A are described in Appendix B, paragraphs a - g respectively. Steps 4260, 4270, 4280, and 4290 of Fig. 64A are described in paragraph h of Appendix B.

Reference is now made to Figs. 64B and 64C

which are simplified flowchart illustrations of the method of Appendix A.

Reference is now made to Figs. 65 and 66. Fig. 65 is a simplified flowchart illustration of subscriber unit operations during a hand-off process provided in accordance with a preferred embodiment of the present invention. Fig. 66 is a simplified flowchart illustration of base station operations during the hand-off process.

In process 5010, the subscriber unit performs a monitoring operation in order to detect a hand-off condition in which the subscriber unit is approaching a fringe area between the old sector through which he is now traveling and a new sector.

Typically, in the present invention, when it is necessary to determine whether a subscriber unit is in a fringe area or not, relative signal strength of control channels in the two sectors is used as a measure of location. Typically, the ratio of the signal strengths, or their difference in dB, is used. Typically, if the power of the control channel of a sector neighboring the current sector is at least 4 times as great as the power of the control channel of the current sector, this is taken to indicate presence in a fringe area and is taken to constitute a hand off condition.

When a hand-off condition is detected (process 5020), the subscriber unit sends a hand-off request, identifying the new sector, to a radio servicing the old sector. The hand-off request is forwarded to a radio servicing the new sector via a base station communicating with all sector radios which provides central services to all sectors (process 5030).

The term "traffic channel key", as used in the present specification, refers to a packet of information allowing a subscriber unit to transmit and receive, the packet preferably comprising one or more transmission frequencies and one or more reception frequencies. In the

case where a plurality of frequencies is provided for either or both of transmission and reception, the traffic channel key preferably also comprises a schedule associating each of the plurality of frequencies with one or more slots, such that the subscriber unit is allowed to broadcast and receive on a specified channel in each slot.

If a traffic channel key is available in the new sector, the new sector will select and transmit a traffic channel key to the subscriber unit. Otherwise (process 5040), the new sector will transmit a "wait signal", also termed herein a "camp-on signal". If a "camp-on signal" is received (process 5050), the subscriber unit continues using the traffic channel key assigned to him by the old sector radio.

If a "camp-on signal" continues to be received when the subscriber unit, is already sufficiently deep into the new sector so as to interfere with other subscribers, the subscriber unit preferably disconnects itself. Typically, a subscriber unit is taken to be sufficiently deep into the new sector when the received power of the control channel in the new sector is significantly stronger than the received power of the control channel in the old sector as, for example, at least 10 times as strong.

If the subscriber unit does receive a traffic channel key from the new sector radio (process 5040), the subscriber unit determines the timing and amplitude of the new sector's downlink (process 5060).

In process 5070, the subscriber unit sends a new sector synchronization signal to the new sector radio which includes information on uplink timing and amplitude. Upon receipt of the new sector synchronization signal, the new sector radio informs the old sector radio that the old sector traffic channel key is no longer required by the subscriber unit and can be reassigned to

another subscriber unit in the old sector (Fig. 66, process 5110).

The subscriber unit then switches from the old sector radio to the new sector radio (process 5080), including: terminating use of the old sector traffic channel key; switching the subscriber unit's downlink timing and amplitude from those of the old sector to those of the new sector, as determined in process 5060; and initiating use of the new sector traffic channel key.

The base station unit receives the hand-off request sent by the subscriber unit in process 5030 and instructs the new sector radio (process 5090) to try and assign a traffic channel key to the subscriber unit, which is not always possible since traffic channel keys are a limited resource in each sector.

If the new sector radio succeeds in assigning a traffic channel key to the subscriber unit undergoing hand-off, the base station, which typically includes a switchboard such as a PABX, sets up a 3-way conference call between the subscriber unit, the old sector radio and the new sector radio (process 5100) to "cover" for communication difficulties while the subscriber unit switches from one sector radio to the other.

The subscriber unit starts operating in the new sector (process 5115). For the new sector radio, there are typically no special timing considerations for broadcasting to the subscriber unit, and standard methods are therefore employed. For proper reception by the new sector radio, it is necessary to utilize correct timing information. The new sector radio has already received a synchronization signal from the subscriber unit (process 5070, above). The synchronization signal is preferably used to provide estimated timing data, and initial reception timing is determined on the basis of the synchronization signal.

Once hand-off has been effected, preferably

including initiation of use of the new sector traffic channel key, terminating use of the old sector traffic channel key and transmission of a synchronization signal from the subscriber unit to the new sector radio, the base station terminates the 3-way conference (process 5120).

The subscriber unit disconnects itself if the subscriber unit is deep into the new sector and has not yet received a traffic channel key from the new sector radio (process 5050). Optionally, self-disconnection also occurs if a traffic channel key is available however no conference bridge is available at the base station such that no 3-way conference call can be set up between the new and old sector radios, and the subscriber unit which is being handed off.

In process 5080, the base station switches the subscriber unit from one sector to another.

Reference is now made to Figs. 67 and 68. Fig. 67 is a simplified flowchart illustration of subscriber unit operations during a hand-off process provided in accordance with an alternative preferred embodiment of the present invention. Fig. 68 is a simplified flowchart illustration of base station operations during a hand-off process provided in accordance with an alternative preferred embodiment of the present invention.

The method of Fig. 67 is similar to the method of Fig. 65, except that hand-off occurs separately on the uplink channel and on the downlink channel. It is appreciated that the hand-off on the two channels may occur in either order, with the uplink hand-off first or with the downlink hand-off first. In Fig. 67, the channels are referred to as the first channel and the second channel.

In Fig. 67, process 5080 of Fig. 65 is replaced with processes 5130, 5140, 5150, and 5160. In process 5130, the subscriber unit waits until no voice

activity is heard on the first channel, preferably by using voice activity detection (VAD), as is well known in the art. In process 5140, the subscriber unit switches to the new sector radio for the first channel only, using a process similar to that described above with reference to process 5080, except that the switch occurs on the first channel only.

In process 5150, the subscriber unit waits until no voice activity is heard on the second channel, preferably by using voice activity detection (VAD), as is well known in the art. In process 5160, the subscriber unit switches to the new sector radio for the second channel only, using a process similar to that described above with reference to process 5080, except that the switch occurs on the second channel only.

Fig. 68 is similar to Fig. 66, with process 5110 of Fig. 66 replaced with processes 5130, 5170, 5150, and 5180. Processes 5130 and 5150 are described above with reference to Fig. 67. Process 5170 is similar to process 5110, except that process 5170 occurs on the first channel only and with a single channel key, the key associated with the first channel, being freed. Process 5180 is similar to process 5170, except that process 5180 occurs on the second channel and not the first channel.

Reference is now made to Fig. 69 which is a simplified block diagram of a power control system for use in a radio communication system, the power control system being constructed and operative in accordance with a preferred embodiment of the present invention. The power control system of Fig. 69 comprises a first station 6100 and a second station 6110.

Although two stations are shown in Fig. 69, it is appreciated that the present invention may be useful in many different kinds of radio communications systems, including a radio communication system comprising any number of stations. In addition, it is appreciated that

the power control system of the present invention may be used to control and optimize transmission power in any radio communication system, with goals including optimizing reception and minimizing interference between stations while minimizing the power used for transmission.

As described herein by way of example, the present invention is described as being used in a mobile radio system. The power control system of the present invention is particularly suited to controlling power in a mobile radio system, since the mobility of stations in a mobile radio system requires relatively frequent power adjustment in order to achieve the goals described above. It is appreciated, however, that the present invention is not limited to use in a mobile radio system, but may be used in any suitable radio communication system.

The power control system of the present invention is particularly suited to controlling power in a multiple access radio communication system, and in a frequency hopping system. It is appreciated, however, that the present invention is not limited to use in multiple access and/or frequency hopping radio communication systems, but may be used in any suitable radio communication system.

As described herein by way of example within a mobile radio system, the first station 6100 may comprise a subscriber unit, typically a mobile station, while the second station 6110 may comprise a base station, typically a fixed station. It is appreciated, however, that the first station and second station, as described herein with regard to various embodiments of the present invention, may be any of a number of types of stations.

The first station 6100 comprises a first station transmitter 6120, which may be any suitable type of radio transmitter, depending on the type of radio communication supported by the radio communication

system. Suitable transmitters include the capability of having their transmission power regulated in response to an external control.

Suitable transmitters include transmitters used in SMR (special mobile radio) and cellular systems, including transmitters suitable for TDMA (time division multiple access), FDMA (frequency division multiple access), AMPS (analog FM), and FHMA (frequency hopping multiple access) systems. One example of a suitable transmitter is the CR-920 Cellular Radio Transmitter Driver, commercially available from Frequency Products, Inc., 3475-M Edison Way, Menlo Park, CA 94025.

The first station 6100 further comprises a power level controller 6130. The power level controller 6130 is operative to control the transmission power of the first station transmitter 6120. The power level controller 6130 may be any suitable power level controller, such as a gain controlled amplifier or a voltage controlled attenuator operatively associated with a fixed gain amplifier, or a digitally controlled step attenuator operatively associated with an amplifier, typically a fixed gain amplifier. One example of a suitable power controller is the RF2410 UHF Programmable Attenuator, commercially available from RF Micro Devices, 7341-D West Friendly Ave., Greensboro, NC 27410.

The first station 6100 also comprises a signal processor 6135, which may be any suitable signal processor as, for example, a DSP 2111, commercially available from Analog Devices.

The first station 6100 also comprises a first station receiver 6140, which may be any suitable type of radio receiver, depending on the type of radio communication supported by the radio communication system.

Suitable receivers include receivers used in SMR (special mobile radio) and cellular systems,

including receivers suitable for TDMA (time division multiple access), FDMA (frequency division multiple access), AMPS (analog FM), and FHMA (frequency hopping multiple access) systems. One example of a suitable receiver is the CR-910 Cellular Radio Dual Conversion Receiver, commercially available from Frequency Products, Inc., 3475-M Edison Way, Menlo Park, CA 94025.

The second station 6110 comprises a second station receiver 6150, which may be any radio receiver suitable to receive the transmissions of the first station transmitter 6120. Radio receivers suitable for the first station receiver 6140 are also suitable for the second station receiver 6150.

The second station 6110 further comprises a power level detector 6160, suitable for detecting the power level of signals received by the second station receiver 6150. The power level detector 6160 may be any suitable power detector, implemented in hardware, in software, or in a combination of hardware and software. Suitable power level detectors include: an AM detector combined with an automatic gain control current state detector.

The power level detector 6160 may alternatively be operative to detect a characteristic other than power, which characteristic provides an alternative measure of signal strength. Examples of such alternative power level detection methods include: computation of signal to noise ratio such as computation of bit energy to noise density ratio in the received signal; computation of carrier to interference ratio; bit error rate performance; channel state performance; frame error rate performance, measured by the rate of bad CRC indications.

The second station 6110 further comprises a power level comparator 6170. The power level comparator 6170 may be any suitable analog or digital power level comparator, typically implemented in software. In

comparing power levels, the power level comparator 6170 is operative to utilize a threshold value appropriate to the particular power detection method used by the power level detector 6160, based on overall detection and decoding performance of the system as a whole. For example, an appropriate threshold value may be approximately 8 dB.

The second station 6110 further comprises a second station transmitter 6180. The second station transmitter 6180 may be any radio transmitter suitable to send transmissions capable of being received by the first station receiver 6140. Radio transmitters suitable for the first station transmitter 6120 are also suitable for the second station transmitter 6180.

It is appreciated that the first station transmitter 6120 and the second station transmitter 6180 may be of different types, including types of transmitters operating on different bands. It is also appreciated that the first station receiver 6140 and the second station receiver 6150 may be of different types, including types of receivers operating on different bands.

The operation of the apparatus of Fig. 69 is now briefly described. Reference is now additionally made to Fig. 70A, which is a simplified flowchart illustrating the operation of the apparatus of Fig. 69. The steps of the method of Fig. 70A preferably include the following:

STEP 6190: Choose initial transmitted power level for first station. The power level controller 6130 sets an initial transmitted power level for the first station 6100. The initial transmitted power level may be set arbitrarily. Typically, the initial transmitted power level may be the maximum power level.

Alternatively, the initial transmitted power level may be based on an open loop estimate of the

required power level, relying on the duality of paths between the first station 6100 and the second station 6110. That is, the expected power loss on the path from the first station 6100 to the second station 6110 is approximately equal to the power loss from the second station 6110 to the first station 6100. This principle is called "path loss duality".

A nominal initial transmitted power level estimate is made based on the expected power loss and the desired received power at the second station. Then, the initial power level is set accordingly, preferably including a safety margin, typically of 10dB over the nominal estimate.

STEP 6192: Transmit first message from first station 6100 to second station 6110. The first station 6100 transmits a first message 6182, shown in Fig. 69, using the first station transmitter 6120 operating at the initial power level, to the second station 6110.

STEP 6194: Receive first message at second station. The first message 6182 is received by the second station receiver 6150.

STEP 6196: Detect received power level at second station. The power level detector 6160 determines the power level of the first message 6182 as received by the second station 6110. As described above with reference to Fig. 69, any number of different methods may be used by the power level detector 6160 to determine the power level of the first message 6182.

STEP 6197: Compare received power level to predetermined value and determine difference. The power level comparator 6170 compares the power level detected by the power level detector 6160 to a predetermined power level.

Typically, power levels and differences in power levels are measured on a dB scale, which is a ratio scale, or as a function of dB level. An arithmetic

difference on a dB scale represents a ratio between two power levels. The term "difference", as used throughout the specification and claims to refer to power levels, refers to any appropriate measure of difference in power as, for example, a difference on the dB scale representing a ratio between two power values.

The predetermined power level represents a sufficiently high received power level at the second station 6110 to allow nominal performance. The predetermined power level is thus dependent on the particular characteristics of the second station receiver 6150. Typically, the predetermined power level may be between approximately 8 dB and approximately 15 dB. The predetermined power level may typically be determined in advance based on the characteristics of the second station 6110. The power level comparator 6170 determines the difference between the predetermined power level and the detected power level.

Typically, the received power level at the second station 6110 may vary irregularly from message to message. It is typically not desired to vary the transmitted power level at the first station 6100 more than is necessary to cause the received power level at the second station 6110 to remain within predetermined tolerances. The actual tolerances depend on the particular characteristics of the communication system of which the first station 6100 and the second station 6110 are a part. Typically, the tolerance is approximately 10 dB.

Therefore, comparing the received power level to the predetermined value preferably includes "smoothing" the received power level to remove such irregular variations. Typically, smoothing is achieved by low pass filtering of the received power level. Examples of appropriate low pass filters include: an FIR filter -- finite impulse response; an IIR filter --

infinite impulse response.

STEP 6201: Transmit second message from second station to first station, including indication of difference. The second station transmitter 6180 transmits a second message 6184, shown in Fig. 69, to the first station. The second message 6184 includes an indication of the difference between the predetermined power level and the detected power level. The indication of the difference may comprise the difference itself or any function thereof.

It is typically desired to minimize the number of messages sent by any station. In order to minimize the sending of unnecessary messages, the second message 6184 preferably also includes other information that would in any case need to be sent from the second station 6110 to the first station 6100, so that no additional message need be sent. Alternatively, in a communication system comprising a control channel intended to carry messages containing system control information, the second message 6184 may optionally be sent on the control channel.

STEP 6202: Receive second message at first station. The first station receiver 6140 receives the second message 6184 and transmits, within the first station 6100, the indication of the difference between the predetermined power level and the detected power level to the power level controller 6130.

STEP 6203: Filter the received message using an appropriate filter, as, for example, a filter implemented in software in signal processor 6135. Reference is now additionally made to Fig. 70B, which is a simplified electronic circuit diagram illustrating the operation of a preferred implementation of step 6203 of Fig. 70A. Fig. 70B illustrates in detail the method of operation of one possible example of an appropriate filter. It is appreciated that many other appropriate

methods of filtering are also possible.

The method of Fig. 70B takes an unfiltered input PWR_CNT_SYM and produces a filtered output HPCS. The unfiltered input PWR_CNT_SYM is a control signal for the method of Fig. 70B. Typically, PWR_CNT_SYM is produced by an automatic gain control process. One possible methodLeor producing PWR_CNT_SYM is as follows: SGCS0 and SCGS1, respectively, be gain control signals from two different reception channels, or diversity channels, 0 and 1. The values of SGCS0 and SGCS1 vary between 0 and 1, with 0 representing minimum signal and maximum gain, and 1 representing maximum signal and minimum gain. Typically, a plurality of previous values as, for example, eight values of SGCS0 and SGCS1 from a current slot and the previous 7 slots, are used;

In each period delta-T:

let k be the index of the current slot;

set AVE_SGCS = (1/16) * SUM (SGCS0(k-i) + SGCS1(k-i)) , where SUM represents a sum from i = 0 to i = 7;

set PWR_CNT_SYM = 0;

if AVE_SGCS < T1, then PWR_CNT_SYM = +1;

if AVE_SGCS > T2, then PWR_CNT_SYM = -1,

where T1 and T2 represent threshold values. Typical threshold values are T1 = 18 * RES and T2 = 22 * RES. A typical value of RES, throughout the method of Fig. 70B, is 1/70. Typically, delta-T has the value 6160 msec for the first 40 iterations and 480 msec afterwards.

As described here, the method of producing PWR_CNT_SYM uses a non-smooth function. It is appreciated that many other methods of producing PWR_CNT_SYM are also possible, such as using a linear function or a non-linear but smooth function.

The method of Fig. 70B is self-explanatory, except as follows:

DZ 6198 is a dead zone, whose output Y depends

on its input X according to the relationship:

$$y = 0 \quad \text{if} \quad |x| < T$$
$$y = +1 \quad \text{if} \quad x > T$$
$$y = -1 \quad \text{if} \quad x < -T \ ;$$

Limiter 6199 limits the signal input between 0 and 1;

dB_to_V 6200 produces the filtered output signal HPCS and, in one preferred embodiment of the present invention, typically operates as follows:

if SPCS < 0.0453 then HPCS = 1 - 3.835 · SPCS,

else if SPCS < 0.144 then HPCS = 0.8 - (SPCS - 0.0627) · 0.76845,

else HPCS = 0.725 - 0.3 · (SPCS - 0.172),

if HPCS < 0.475 then HPCS = 0.475;

typical values for the parameters α, β, and τ are defined as follows:

for the first N0 iterations, with N0 typically equal to 30: $\alpha = 1$, $\beta = 0$, and $\tau = 5 * \text{RES}$,

for the next N1 iterations, with N1 typically equal to 10: $\alpha = 1$, $\beta = 0$, and $\tau = 2 * \text{RES}$,

afterwards: $\alpha = 0.1$, $\beta = 0.5$, and $\tau = 0.4 * \text{RES}$; and

the parameters IC_X and IC_SPCS represent initial conditions, having typical values of IC_X = 0 and IC_SPCS between 0 and 1, depending on particular implementation details of the filter. Typically, a value of IC_SPCS near 0 indicates transmission of a transmitted signal at the lower end of the dynamic range. Conversely, a value of IC_SPCS near 1 indicates transmission of a transmitted signal at the higher end of the dynamic range.

STEP 6204: Modify transmitted power level for first station. The power level controller 6130 modifies

the current power level based on the initial power level and the difference. Typically, the new power level is set to be the initial power level plus the filtered difference, within a predetermined maximum and minimum power level.

Alternatively, another suitable method for modifying the power level may be used. For example, as described below with reference to Fig. 71, the difference, typically a difference on a dB scale representing a ratio between power levels, may be stored and the modification to the current power level may be based partly on the stored difference. Alternatively, the difference may be compared to a stored minimum or threshold value and the current power level may be modified only if the difference exceeds the stored threshold value. Typically, the stored threshold value may be a predetermined value such as, for example, 5dB or 10dB.

Preferably, as subsequent messages are transmitted, the method of Fig. 70A may be performed iteratively, although the method of Fig. 70A may be performed only once. In the case of iteration, the power level is modified repeatedly over time in order to adapt to changing needs. Iteratively modifying the power level is particularly preferred in the case where at least either the first station 6100 or the second station 6110 is a mobile station, so that the positions of the first station 6100 and the second station vary over time, as do the distance between the first station 6100 and the second station 6110, and the locations of the first station 6100 and the second station 6110 relative to sources of interference and of shadowing.

Reference is now made to Fig. 71, which is a simplified flowchart illustrating the operation of step 6204 of Fig. 70A according to an alternative embodiment of the present invention. As described above with

reference to Fig. 70A, Fig. 71 illustrates the alternative wherein the difference between the predetermined power level and the detected power level is stored and the modification to the current power level is based partly on the stored difference.

The method of Fig. 71 preferably includes the following steps:

STEP 6206: Store indication of filtered difference. The indication of the filtered difference between the predetermined power level and the detected power level is stored for future retrieval. The filtered difference may be stored along with an indication of when the difference occurred, such as, for example, the time or a message identification. Alternatively, some function based on the filtered difference may be stored, such as, for example, the sum of all filtered differences until the current filtered difference.

STEP 6208: Choose increment based on both current difference and stored difference. The increment to be used in modifying the current power level is chosen based on both the current filtered difference and on one or more stored filtered differences.

Typical methods for choosing an increment include: computing an average of the current difference and the stored difference and using the average as the increment; computing a trend of stored differences and the current difference and choosing the sign of the increment based on the trend; and computing the magnitude of the increment as a function of the current difference and the stored differences and also applying a stored threshold difference and choosing a non-zero increment only if the computed function exceeds the stored threshold difference. Typically, the stored threshold difference may be a predetermined value such as, for example 5dB or 10dB.

Reference is now made to Fig. 72 which is a

simplified block diagram of a power control system for use in a radio communication system, the power control system being constructed and operative in accordance with an alternative preferred embodiment of the present invention.

The power control system of Fig. 72 comprises a first station 6210. Except as described below, the first station 6210 is similar to the first station 6100 of Fig. 69. The power control system of Fig. 72 also comprises a second station 6220. Except as described below, the second station 6220 is similar to the second station 6110 of Fig. 69.

Although two stations are shown in Fig. 72, it is appreciated that the present invention may be useful in many different kinds of radio communication system, including a radio communication system comprising any number of stations. In addition, it is appreciated that the power control system of the present invention may be used to control and optimize transmission power in any radio communication system, with goals including optimizing reception and minimizing interference between stations while minimizing the power used for transmission.

The first station 6210 comprises a first station transmitter 6120, a first station receiver 6140, and a signal processor 6135, each as described above with reference to Fig. 69. The first station 6210 need not comprise other elements of the first station 6100 as described above with reference to Fig. 69.

The second station 6220 comprises a second station receiver 6150, a power level detector 6160, and a second station transmitter 6180, each as described above with reference to Fig. 69.

The second station 6220 further comprises a power level comparator 6230. The power level comparator 6230 may be similar to the power level comparator 6170 of

Fig. 69.

The second station 6220 further comprises a power level controller 6240. The power level controller 6240 is operative to control the transmission power of the second station transmitter 6180. The power level controller 6240 may be similar to the power level controller 6130 of Fig. 69.

The operation of the apparatus of Fig. 72 is now briefly described. Reference is now additionally made to Fig. 73, which is a simplified flowchart illustrating the operation of the apparatus of Fig. 72. The steps of the method of Fig. 73 include the following:

STEP 6243: Determine desired received power level at first station 6210. The desired received power level represents the optimum received power level at the first station 6210. The desired received power level may typically be determined in advance based on the characteristics of the first station receiver 6140. The desired received power level may be a predetermined parameter known to the second station 6220, or may alternatively be communicated to the second station 6220, either before the operation of the method of Fig. 72 or as part of a message transmitted from the first station 6210 to the second station 6220.

STEP 6244: Transmit signal from first station 6210 to second station 6220, including indication of first station 6210 transmitted power level. The first station 6210 transmits a signal comprising a first message 6241, shown in Fig. 72, using the first station transmitter 6120 operating at a transmitted power level, to the second station 6220. The first message 6241 includes an indication of the transmitted power level and an indication of the noise level received by the first station 6210 in previous transmissions from the second station 6220. Typically, the portion of the first message 6241 comprising an indication of the transmitted

power level and the received noise level may be transmitted on a different channel than the remaining portion of the first message 6241. Such a different channel is typically called a control channel. In addition to the indication of the transmitted power level, the transmissions on the control channel may typically comprise other signals useful for control of the radio communication system.

STEP 6246: Receive signal at second station. The signal comprising the first message 6241 is received by the second station receiver 6150.

STEP 6248: Measure received power level of signal at second station 6220. The power level detector 6160 determines the power level of the first message 6241 as received by the second station 6220, as described above with reference to Figs. 69 and 70.

STEP 6250: Compare received power level to transmitted power level and compute transmission loss. The power level comparator 6230 compares the power level detected by the power level detector 6160 to the transmitted power level.

Typically, the transmission loss is computed by computing the difference between the received power level and the transmitted power level.

The transmitted power level was received by the second station 6220 as part of the first message 6241. The power level comparator 6230 computes the difference between the transmitted power level and the detected power level and thus determines the transmission loss between the first station 6210 and the second station 6220.

The method of Fig. 73 relies on the assumption of path loss duality, as explained above with reference to Fig. 70A. In the case of the apparatus of Fig. 73, the computed path loss of the first message 6241 sent from the first station 6210 to the second station 6220 is

taken, according to path loss duality, to be the expected value of transmission loss for a second message 6242 sent from the second station 6220 to the first station 6210.

Typically, the received power levels at the second station 6220 and at the first station 6210 may vary irregularly from message to message. It is typically not desired to vary the transmitted power level at the second station 6220 more than is necessary to cause the received power level at the first station 6210 to remain within acceptable limits. Depending on the characteristics of the first station receiver 6140, the limits may typically be between 10dB and 70dB. Therefore, comparing the received power level to the transmitted power level preferably includes smoothing the received power level to remove such irregular variations. Alternatively, smoothing may be done in the first station 6210, typically in the power level controller 6130, as described above with reference to Fig. 70A.

STEP 6252: Determine transmitted power level for second station 6220 based on desired received power level, on transmission loss, and on received noise level at the first station 6210. The transmitted power level for the second station 6220 is computed based on the desired received power level, determined in advance; the expected transmission loss in the direction from the second station 6220 to the first station 6210, determined in step 6250; and the received noise level at the second station, also determined in step 6250. Typically, the transmitted power level for the second station 6220 is the sum in dB of the desired received power level, the expected transmission loss, and the received noise level at the second station.

The second message 6242 is then sent with the transmitted power level. Typically, the transmitted power level is computed by signal processor 6135 by

computing the sum of the desired received power level and the expected transmission loss.

Reference is now made to Fig. 74, which is a simplified block diagram of a power control system for use in a radio communication system, the power control system being constructed and operative in accordance with another alternative preferred embodiment of the present invention. The apparatus of Fig. 74 comprises the components of station 6110 of Fig. 69 and of station 6220 of Fig. 72. Message 6254 comprises message 6184 of Fig. 69 and message 6241 of Fig. 72, while message 6256 comprises message 6182 of Fig. 69 and message 6242 of Fig. 72.

Reference is now made to Fig. 75, which is a simplified flowchart illustrating a preferred method for operating the apparatus of Fig. 74. The method of Fig. 75 combines the methods of Figs. 70A and 73. In the method of Fig. 75, the open loop control method of Fig. 73 is taken as the master, while the closed loop control method of Fig. 70A is used to adjust the results of the open loop method. This is done because, typically, the closed loop method is slower but more accurate.

It is appreciated that the apparatus of Fig. 74 is symmetrical, that is, either station 6110 or station 6220 of Fig. 74 may take the role of first station or second station in the methods of Figs. 70A and 73.

First, the open loop power control method of Fig. 73 is performed (step 6270). Concurrently, the closed loop power control method of Fig. 70A is performed (step 6280). Finally, the results of the open loop method are adjusted based on the results of the closed loop method (step 6290).

Reference is now made to Fig. 76, which is a simplified partly pictorial, partly block diagram illustration of a radio communication system constructed and operative in accordance with a preferred embodiment

of the present invention. The system of Fig. 76 comprises a base station (BS) 7010 and a subscriber unit (SU) 7020. Typically, the SU 7020 is a mobile subscriber unit.

It is desirable, particularly in a slotted radio communication system, for messages to be sent at a pre-assigned time, as, for example, at the time specified for a slot assigned to the sending station. When time ~~alignment of a message is significantly in error,~~ problems arise since the message is sent outside of its assigned slot.

The operation of the system of Fig. 76 is now briefly described. A regular transmission 7030 is sent from the SU 7020 to the BS 7010. The BS 7010 determines the time alignment error of the regular transmission 7030 and sends a timing alignment message 7040 to the SU 7020. The SU 7020 then corrects its timing for subsequent transmissions based on the timing alignment message 7040.

It is appreciated that the time alignment of messages sent from the SU 7020 generally varies over time due to a number of factors as, for example, movement of the SU 7020 relative to the BS 7010.

Reference is now made to Fig. 77, which is a simplified flowchart illustration of a method for time alignment in the radio communication system of Fig. 76.

Initially, when the SU 7020 first receives a control channel signal sent by the BS 7010, as, for example, when the SU 7020 is first switched on or, for example, when the SU 7020 moves into the range of another base station and first receives a control signal therefrom, the SU 7020 resets any time alignment data which it may have accumulated. The reason for resetting the time alignment data in those circumstances is that time alignment relates to alignment with a particular base station, so that being switched on or coming into the range of a new base station implies that the old time

alignment data is invalid.

The SU 7020 transmits an uplink transmission (process 7100), which may comprise, for example, an access channel transmission (ACH) or an uplink traffic channel transmission (UTCH). An ACH is a uplink control/information transmission sent from the SU 7020 to the BS 7010. An uplink traffic channel transmission is a regular uplink message, such as a message carrying voice or data. The uplink transmission is transmitted via an air interface to the BS 7010.

In the BS 7010, the uplink transmission is detected and decoded (process 7120). Detection and decoding comprises determining the timing offset (D) of the uplink transmission, that is, the difference between the time at which the uplink transmission is received and the assigned time for that uplink transmission.

A determination is made as to whether the timing offset D is within an acceptable range. Preferably, the absolute value of the timing offset D is compared to a threshold offset D_0 (process 7130). Preferably, the threshold offset D_0 is set at 1/2 of the time necessary to transmit a single symbol.

In the case where the absolute value of the offset timing is less than the threshold offset, normal transmission processing continues and no time alignment correction is made.

In the case where the absolute value of the offset timing is greater than the threshold value, a timing alignment command TA_COM(D) indicating the time offset value D, is transmitted (process 7135) from the BS 7010 to the SU 7020 via the air interface.

The timing alignment command TA_COM(D) is received at the SU 7020. Preferably, the SU 7020 immediately acknowledges the receipt of the timing alignment command to the BS 7010 (acknowledgement not shown in Fig. 77). The SU 7020 corrects its time

alignment in accordance with the information received in the timing alignment command TA_COM(D) (process 7140).

Preferably, the timing alignment command TA_COM(D) comprises a unique sequence number, and the SU 7020 checks the sequence number and does not perform a time alignment correction if a message with a duplicate sequence number is received.

A preferred implementation of correction of time alignment in process 7140 is as follows.

Each time SU 7020 receives a TA_COM(D) command, it immediately acknowledges receiving the command. SU 7020 then shifts its local transmitter timing (T_SLT) by 2*D increments, in the opposite direction, where D = -31..32. This parameter shifting is performed as follows:

Gradually shifting the transmitter timing by one increment of 1/16 symbol during each period of 480 msec time interval between Uplink allocation and deallocation. For all other cases, the operation is immediate.

Preferably, a second timing alignment message is not sent until a predetermined delay after acknowledgment of the first timing message. Typically, the predetermined delay is approximately 60 seconds.

Preferably, certain messages from the SU 7020 which, if acted upon, might interfere with system operations, are ignored if the SU 7020 is found not to be time aligned. Examples of such messages preferably include messages requesting allocation of a channel, such as, for example, traffic and access channel allocation.

Preferably, the BS 7010 will ignore messages associated with the SU 7020 if the SU 7020 is not operating in the sector of the BS 7010, or containing a code not suited to be sent by a subscriber unit.

Preferably, the timing alignment command TA_COM(D) comprises an indication of whether the timing alignment is based on an access channel transmission or a

traffic channel transmission. Preferably, the SU 7020 compares the message type, access or traffic, to the message most recently sent by the SU 7020 and corrects its time alignment only if the message type contained in the timing alignment message is identical to the type of the most recent transmission.

A preferred method for determining the timing offset D in step 7120 is now described.

For each BS 7010 BACH (Basic Access Channel) transmission correctly received, that means a message with Good CRC (Cycle Redundancy Check), the CA_FP (Control Access Frame Processor) picks up its BACH_DLY (BACH Delay) command and set it equal to D.

When the TCH_FP (Traffic Channel Frame Processor) detects either two adjacent Voice/Data frames, each with Good CRC or a TCHM (Traffic Channel Message) with Good CRC, at the same time with D_ERR=0, it chooses the current HDCS (Hardware Delay Control Signal) and sets it equal to D. It should be noted that the parameter D_ERR should be available. A preferred method for implementing steps 7130 and 7135 is as follows.

For each new D, check whether $D > D_0$. If the answer is affirmative, than a TA_CMD(D) with the D parameter and sequence number is sent to SU 7010. D is given a range compatible with the number of bits available. The TA_CMD(D) is then acknowledged.

Preferably, the SU 7020 transmits a periodic status message to the BS 7010. Thus, an opportunity to perform the method of Fig. 77 will arise periodically even when no data message is otherwise transmitted by the SU 7020 to the BS 7010.

Preferably, the SU 7020 accumulates the set of all time alignment performed since the last reset was done upon receipt of an initial time alignment command. A detailed implementation of the handling of accumulated time alignment is as follows.

After each timing reset (SU 7020 resets its transmitter timing (T_SLT) after completing time acquisition) or sector switching (Handoff and/or reserved sector) SU 7020 should transmit its Tx Timing Status (During regular operation (IDLE, VOICE, etc.)), the Subscriber accumulates the net Tx Timing Shift and subtracts it from the current Rx timing, i.e. the sum of all Tx shifts minus the sum of all Rx shifts since the last reset. The current accumulation result is noted by Tx Timing Status. The shifting operations, defined hereinabove, are limited so that the Tx Timing Status does not exceed +/- 96 shifting increments of 1/16 symbol) to the new sector and wait for an acknowledgement. The transmission has up to 1024 possible values and thus the message should include 10 bits CACH (Composite Access Channel)transmission or UTCH_IBOH (Uplink Traffic Channel Inband Overhead).

If any SU_STATUS signals are transmitted during a time interval, described hereinabove, (i.e. from receiving a TA_CMD(D) command until the end of the time shifting) then the SU_STATUS transmission is delayed until after the end of this interval.

If any SU_STATUS is transmitted during any event which requires Tx Timing Status transmitting (Reset, Handoff, etc.) then the SU_STATUS signal is delayed until the Tx Timing Status is transmitted and acknowledged.

Using the time alignment data, the BS 7010 may optionally compute the range of the SU 7020. A preferred method for computing the range of an individual subscriber unit is as follows:

Each time BS 7010 needs the range of any eligible Subscriber Unit, the range is evaluated by the following:

$$R = (D + SUM(D_i) + D_0/2) *C *T_0/2 + R_0$$

where R is Range of the Subscriber Unit;

D is the last BACH_DLY or the current SDCS (Software Delay Control Signal) if within a UTCH receiving;

D_i is the BACH_DLY or the SDCS parameter of the TA_CMD(D) number i;

TA_NUM is the number of TA_CMD(D) which were sent and acknowledged to this Subscriber Unit, since the last "Tx Timing Status";

D_0 is the last "TX Timing Status" of the Subscriber Unit;

C is the speed of light (~3.0E+08 m/sec);

T_0 = 6.78 microsec;

R_0 is a calibration parameter (sector dependent).

and the SUM(D_i) is taken from 1 to TA_NUM.

Reference is now made to Fig. 78, which is a simplified partly-pictorial, partly block-diagram illustration of a radio communication system constructed and operative in accordance with a preferred embodiment of the present invention. The system of Fig. 78 comprises a transmitting station 8050 and a receiving station 8060, both of which may be any appropriate type of radio communication station, including stations of types which are well-known in the prior art. Typically, the stations are part of a frequency hopping multiple access (FHMA) communication system.

The operation of the system of Fig. 78 is now briefly described. The transmitting station 8050 breaks a message into a plurality of sub-messages and transmits the sub-messages to the receiving station 8060 in sub-message transmission 8070. The receiving station receives each sub-message, checks the sub-message for errors and, if the sub-message contains an error, adds the sub-message to a list of bad sub-messages.

When all of the sub-messages are received by

the receiving station 8060, the receiving station 8060 preferably transmits a retransmission request 8080 to the transmitting station 8050, comprising a list of sub-messages which were not received correctly. The transmitting station 8050 then retransmits the requested sub-messages in a retransmission 8090 to the receiving station 8060.

Alternatively, the receiving station 8060 may transmit an acknowledgement message to the transmitting station 8050, comprising an acknowledgement that all sub-messages were received correctly. The absence of such an acknowledgement is typically taken as the equivalent of a retransmission request for all sub-messages.

Reference is now made to Fig. 79, which is a simplified block diagram illustration of a preferred method for operating the system of Fig. 78. The method of Fig. 79 preferably includes the following steps:

STEP 8100: Break message into sub-messages. The message is broken into sub-messages at the transmitting station 8050. The sub-messages may be of any appropriate size. Preferably, the size of sub-messages is chosen in order to minimize, in practice, the mean or average number of retransmissions.

STEP 8110: Add error detection code to each sub-message. Any appropriate error detection code, as is well known in the art, is added to each sub-message. Preferably, CRC code is used. Alternatively, in place of error detection code other means for detecting errors, such as checking received messages for internal consistency, may be used.

STEP 8120: Transmit sub-message. The current sub-message is transmitted from the transmitting station 8050 to the receiving station 8060.

STEP 8130: Receive, detect, and decode sub-message. The receiving station 8060 receives the sub-message and decodes the sub-message so that the data

portion and the error detection code can be examined.

STEP 8140: Check error detection code of sub-message. The error detection code of the sub-message is checked to see whether any errors are indicated.

STEP 8150: Was sub-message received correctly? Check, preferably based on the error detection code, whether the sub-message was received correctly.

STEP 8160: Mark sub-message bad. If the sub-message was not received correctly, the sub-message is marked bad in an internal list maintained by the receiving station 8060.

STEP 8170: Was last sub-message received? Check whether the sub-message just processed was the last sub-message in the message. If not, processing continues at step 8120, described above.

STEP 8180: Was any sub-message marked bad?

STEP 8190: Acknowledge message. If no sub-message was marked bad, the receipt of the entire message is acknowledged by the receiving station 8060 to the transmitting station 8050. Alternatively, the receiving station 8060 may notify the transmitting station 8050 that certain messages were not received correctly by sending a message comprising a list of sub-messages which were not received correctly.

STEP 8200: Request retransmission. If any sub-message was marked bad, the receiving station 8060 requests the transmitting station 8050 to retransmit those sub-messages which were marked bad. Preferably, only the sub-messages marked bad are retransmitted. Processing then continues at step 8120, described above.

Reference is now made to Fig. 80, which is a simplified block diagram illustration of a preferred error detection method useful in conjunction with the method of Fig. 79. As shown in Fig. 80, the method of Fig. 80 is preferably performed at the completion of the method of Fig. 79.

In the case where the transmitting station 8050 waits to receive an acknowledgement from the receiving station 8060, the transmitting station 8050 may typically retransmit a plurality of sub-messages comprising both sub-messages which were not received correctly and sub-messages which were already received correctly. It is appreciated that, in the case where a list of sub-messages to retransmit is sent from the receiving station 8060 to the transmitting station 8050, it is also possible that the sub-messages retransmitted may comprise sub-messages which were already received correctly. Thus, a given sub-message may be received correctly more than once.

If a sub-message is received correctly, that is, with correct error detection code, more than once, but the actual contents of the sub-message contained in the sub-message is different in different instances of the sub-message (step 8210), the receiving station 8060 preferably determines which sub-message contents to use based on a decision criterion as, for example, based on taking the majority of the different instances of the sub-message or based on taking the most prevalent of the different instances of the sub-message (step 8220).

Reference is now made to Fig. 81, which is a simplified block diagram illustration of another preferred error detection method useful in conjunction with the method of Fig. 79. The method of Fig. 81 is preferably performed at the completion of the method of Fig. 79.

Preferably, an additional error detection code is assigned to the message as a whole and is transmitted from the transmitting station 8050 to the receiving station 8060 either as part of one or more of the sub-messages or as a separate message (step 8230). Preferably, the additional whole message error detection code is more reliable than the error detection codes of

each sub-message. For example, in the case where CRC code is used for error detection, the CRC code of the whole message is computed using more bits than the CRC codes of the individual sub-messages.

The receiving station 8060 checks the whole message error detection code (step 8240). If the whole message error detection code is erroneous, the receiving station 8060 checks each possible combination of sub-messages received, for all sub-messages where more than one instance of the sub-message was received, looking for a combination of sub-message instances which yields correct error detection code (step 8250). If such a correct combination is found, that combination as a whole is taken to be the correct message.

In case no combination yielding correct error detection code is found, the receiving station 8060 requests retransmission of sub-messages for which no clear majority or other indication of definitely correct reception is found (step 8260).

The steps of the method of Fig. 81 are preferably repeated until correct whole message error detection code is found (step 8245). Preferably, the steps are only repeated until some predetermined repetition limit is reached as, for example, a maximum number of retransmission requests or a maximum time to receive a single message.

Referring now to Figs. 82 - 84, a preferred memory ARQ (memory automatic repeat request) method, also termed herein a "MARQ method", is now described.

a. The receiver tests the validity of all frames received based on CRC (cyclic redundancy checking). A frame is considered valid if one of the following conditions apply:

1) It was received only once with a valid CRC;

2) It was received more than once with a valid CRC and at least twice with the same content.

If a data frame is received twice with a valid CRC but with differring content, the frame received first is preferred (Double_Ambiguous_Frames).

If a frame is received three times with a valid CRC but with differring content, the frame received first is preferred, then the second, etc. (Triple_Ambiguous_Frames).

b. If all the received frames are valid, the receiver computes the CS ("check sum").

c. If the CS result is incorrect and the following criterion applies to the number of Ambiguous_Frames of any type:

$$2^{N2} * 3^{N3} <= 8$$

where N2 is the number of Double_Ambiguous_Frames, and N3 is the number of Triple_Ambiguous_Frames, then every possible permutation is tried and the validity of CS tested for each.

d. If after the tests described in b and c, above, are carried out, a correct result is achieved, then the process is completed.

e. If a correct result is not achieved, it is then necessary to wait for retransmission of the message, and after the latter is received, the CS tests described above in b and c are performed.

f. If 4 frames are received with a valid CRC but with differing content, the MARQ procedure is terminated.

g. The application performs a reset when an incorrect response of any kind is received after nine attempts.

Reference is now made to Fig. 82 which is a generalized block diagram illustration of a portion of a subscriber unit in a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present

invention.

A frequency hopping multiple access communication system, in accordance with a preferred embodiment of the present invention, utilizes a frequency hopping multiple access communication network and a multiplicity of base stations, at least some of which receive and transmit information at a plurality of radio frequencies over the frequency hopping multiple access communication network.

The system also includes a multiplicity of subscriber units, each receiving and transmitting information at a plurality of radio frequencies via the frequency hopping multiple access communication network.

At a subscriber unit, a receiving and transmitting unit, generally indicated by reference numeral 9010, is operable to receive and transmit radio-frequency (RF) signals.

Receiving and transmitting unit 9010 preferably includes a first antenna 9012, a second antenna 9014 and a radio unit (RU) 9016. Antennas 9012 and 9014 are operable to establish communication channels with a base station (not shown). Preferably, antennas 9012 and 9014 operate in the frequency range 890 - 950 MHz.

In radio unit 9016, RF signals are received at receivers 9018 and 9020, also referred to as RXD and RX respectively. Receiver 9018 is coupled to antenna 9012 and receiver 9020 is coupled to antenna 9014 via a duplexer unit 9022. Preferably, receivers 9018 and 9020 are converters which convert RF signals to intermediate frequency (IF) signals.

In a transmission mode, only antenna 9014 is employed. In a reception mode however, both antenna 9012 and 9014 are employed to achieve space diversity. In that case, receivers RXD and RX determine the quality of reception of the corresponding received signals and the best of the corresponding received signals are selected

for processing.

Preferably, receivers 9018 and 9020 are coupled to a combined gain and frequency control unit 9024 which is operable to provide separate automatic gain control (AGC) signals and common automatic frequency control (AFC) signals to receivers 9018 and 9020. In an alternative embodiment of the present invention gain and frequency control unit 9024 is not a combined unit but may rather include a separate gain control unit and a separate frequency control unit.

Gain and frequency control unit 9024 is coupled to a synthesizer unit 9026 which is coupled to a transmitter 9028. Transmitter 9028 is coupled to a power control unit 9030. Gain and frequency control unit 9024 provides signals to synthesizer unit 9026 and receives inputs, including a clock signal and data, from a modem 9034 in a baseband unit (BBU) 9032.

Receivers 9018 and 9020 are also coupled to synthesizer unit 9026 which generates signals necessary to downconvert signals received by receivers 9018 and 9020. Receivers 9018 and 9020 provide the downconverted signals of the intermediate frequency (IF) to modem 9034 in BBU 9032.

Synthesizer unit 9026 is also operable to generate signals necessary to modulate signals transmitted by transmitter 9028.

The signals generated by synthesizer unit 9026 to downconvert the signals received by receivers 9018 and 9020, and the signals generated by synthesizer unit 9026 to upconvert the signals modulated at modem 9034 and which are transmitted by transmitter 9028 are preferably hopping signals that are generated in accordance with control signals communicated to and from modem 9034 in BBU 9032. Different hopping signals may be generated for transmission and for reception.

Transmitter 9028 receives control signals from

modem 9034 in BBU 9032 and power control signals from power control unit 9030, which also receives inputs from modem 9034 in BBU 9032. Transmitter 9028 is also coupled to a service board (not shown) which provides electric power to transmitter 9028. Transmitter 9028 outputs data for transmission to antenna 9014 via duplexer unit 9022.

In a preferred embodiment of the present invention modem 9034 includes a DSP (digital signal processing) unit 9038 in which a collision avoiding and channel coordinating algorithm is performed as described herein after. Alternatively, DSP unit 9038 may be a separate unit which is coupled to modem 9034.

Reference is now made to Fig. 83 which is a generalized block diagram illustration of a portion of a base station 9100 in a frequency hopping multiple access communication system constructed and operative in accordance with a preferred embodiment of the present invention.

The base station includes a group of receivers 9102, each coupled to an antenna 9104. Preferably, each receiver in the group of receivers 9102 is operable to receive information signals over one frequency channel.

The group of receivers 9102 is coupled to a group of slot processors (SP) 9106. Each receiver in the group of receivers 9102 is coupled to a separate slot processor in the group of SP 9106.

The group of SP 9106 is coupled to a communication bus 9108, preferably a modified HDLC communication bus.

A group of frame processors (FP) 9110 is also coupled to communication bus 9108. The frame processors in the group 9110 receive inputs from the slot processors in the group 9106 via communication bus 9108.

In high traffic frequency hopping multiple access communication systems there may occasionally be a case in which a group of frequencies is allotted to more

than one subscriber simultaneously, such as to two subscribers. In such a case, which is indicated as collision, communication with one of the subscribers, or with both subscribers, may be disconnected.

Disconnection in a case of collision may occur since slot receivers, in a first subscriber unit, may receive signals which are transmitted to a second subscriber unit and vice versa. Such collisions may be avoided by employing methods as described herein after with reference to Figs. 84 - 86.

Reference is now made to Fig. 84 which is a flow chart illustration describing the operation of a collision avoidance and channel coordinating algorithm employed in the apparatus of Fig. 83.

A plurality of frequency hopping channels for transmitting and receiving information signals are provided to a plurality of subscriber units. At each channel the information is transmitted over slots which are defined in a time domain and in a frequency domain.

When a first subscriber and a second subscriber, located at adjacent sectors, communicate simultaneously, each with a separate third party, collisions may occur which may cause disconnection of the conversations of both subscribers.

In a preferred embodiment of the present invention collisions are prevented by allowing each subscriber unit to skip transmission of at least one slot selected in accordance with a predetermined sequence during transmission of information. Alternatively, the predetermined sequence may be transmitted to each subscriber unit over a control channel.

Although part of the information may be lost it is to be appreciated that most of it may be reproduced to a level of acceptable quality, such as a BER (Bit-Error-Rate) of 1.0E-3.

Since the non-transmitted slots are ordered in

a predetermined sequence, a receiver at a receiving end recognizes a non-transmitted slot and builds an inactive slot to replace the non-transmitted slot by including in the inactive slot a plurality of inactive symbols having imparted a confidence level zero (0). Preferably, a slot includes 38 symbols and the confidence level zero is imparted in an ordered or random sequence.

In accordance with a preferred embodiment of the present invention the system includes a module of error correction which is typically performed by a dedicated ASIC (Application Specific Integrated Circuit) module (not shown).

The module of error correction is operable to apply a minimum weight to the inactive symbols during processing of the slots so as to minimize information disruption.

Forced disconnection of subscribers is performed at a base station in accordance with their location in a fringe area. The location of a subscriber in a fringe area is determined by monitoring the control channels. If the subscriber unit receives signals having similar magnitude from the control channels of two adjacent sectors then the subscriber unit determines that it is in a fringe area. Magnitudes of the two control channels are considered to be similar if the difference between the magnitudes is of the order of 6 dB or less.

In accordance with another preferred embodiment of the present invention collisions may be prevented by using another algorithm which employs a preselected probability to determine whether to transmit over a time slot or not.

Reference is now made to Fig. 85 which is a flow chart illustration describing the operation of another collision avoidance and channel coordinating algorithm which is performed at a base station including a transmitter which transmits to a subscriber unit in a

frequency hopping multiple access communication system.

In accordance with a preferred embodiment of the present invention a transmitter within a first sector transmits to a first subscriber within the first sector. If the first subscriber is located within a fringe area then the transmitter transmits the information to the first subscriber sequentially over all the slots (with a probability 1). If the first subscriber is not in a fringe area then the base station determines, for each of at least one time slot, whether or not to transmit from the transmitter to the first subscriber during each of the at least one time slot.

The determination if the first subscriber unit or a second subscriber unit or both are in the fringe area may be achieved by monitoring the control channels at the first subscriber unit and at the second subscriber unit respectively as mentioned before with reference to Fig. 84.

If the first subscriber is not in a fringe area determination is made at the base station, for at least one time slot, whether, during the time slot, there exists a problematic subscriber associated with a neighboring sector who is located within the fringe area between the first and neighboring sectors and who is subject to interference due to transmission from the transmitter to the first subscriber. If no such problematic subscriber exists, then the transmitter transmits in the time slot.

If there is such a problematic subscriber, then the base station defines, for each of a plurality of time segments, a partition of the time during which the transmission occurs. The base station also determines, for each time segment including at least one time slot, whether or not the number of time slots within the time segment in which transmission did not take place exceeds a threshold number of time slots, for example 2. If the

threshold is exceeded, then the transmitter transmits over each time slot at which the determination is currently performed.

If there are some time slots, less than the threshold, over which transmission did not take place, a sliding time window is defined and determination is made, for each of a plurality of positions of the sliding time window including n>1 time slots, whether or not the number of time slots within the sliding window, as currently positioned, in which transmission did not take place exceeds a threshold number of time slots. If the threshold is exceeded, the transmitter transmits over each time slot which is currently determined.

In a preferred embodiment of the present invention the determination whether to transmit or not is performed by preselecting a fixed probability p<1. Each time a determination whether to transmit or not is required, a random number between 0 and 1 is selected and compared to the preselected probability p. If the selected random number is equal to p or exceeds p then the transmitter transmits over the currently determined time slot.

If the selected random number is less then p then an average number AVE_SS of skipped transmittals over a time period T is computed. If AVE_SS exceeds a preselected allowable constant number X of skipped transmittals, the transmitter transmits over the current time slot. Otherwise, transmittal over the time slot is skipped.

In a preferred embodiment of the present invention the base station may assign time-slots to each of a plurality of subsectors within a first sector at which the subscriber unit is located, and to each of a plurality of subsectors within a neighboring sector.

Each sector may be divided to central and peripheral subsectors such that the same time-slot is

assigned to a peripheral subsector in the first sector and to a central subsector in the neighboring sector. Preferably, the base station may assign more power to downlink transmissions to subscribers within the peripheral subsectors than to downlink transmissions to subscribers within the central subsectors.

The power differences between the subsectors may be employed to prevent collision. In a preferred embodiment of the present invention an air resource may be allocated to the subscribers within the first sector so as to reduce the maximum probability, over the subscribers within the first sector, of existence of a problematic subscriber. Thus, the transmitter may transmit with a probability 1 if the first subscriber is inside the fringe area even if there is a problematic subscriber in a neighboring sector.

Preferably, the air resource may include one of TDMA (time division multiple access) time slots, FDMA (frequency division multiple access) channels frequencies and FHMA (frequency hopping multiple access) time/frequency sequences. In a preferred embodiment of the present invention each of the time slots includes an active time slot.

Reference is now made to Fig. 86 which is a flow chart illustration describing the operation of a collision avoidance and channel coordinating algorithm which is performed at a subscriber unit and is operative in accordance with a preferred embodiment of the present invention. It is to be appreciated that the collision avoidance and channel coordinating algorithm performed at the subscriber unit is substantially similar to the algorithm performed at the base station.

In a preferred embodiment of the present invention a subscriber within a first sector is operable to transmit to a base station. If the subscriber is not located within a fringe area, then the subscriber

transmits to the base station. Otherwise, the subscriber determines, for each of at least one time slot, whether or not to transmit during the time slot.

Preferably, at the subscriber unit definition is made, for each of a plurality of time segments, of a partition of the time during which the transmission occurs, each time segment including at least one time slot, and determination is made whether or not the number of time slots within the time segment in which transmission did not take place exceeds a threshold number of time slots, for example 2. If the threshold is exceeded, the subscriber unit transmits in a current time slot.

If there are some time slots, less than the threshold, over which transmission did not take place, a sliding time window is defined and determination is made, for each of a plurality of positions of the sliding time window including n>1 time slots, whether or not the number of time slots within the sliding window, as currently positioned, in which transmission did not take place exceeds a threshold number of time slots. If the threshold is exceeded, the subscriber unit transmits over each time slot which is currently determined.

As in the algorithm performed at the base station, the determination whether to transmit or not is performed by preselecting a fixed probability p<1. Each time a determination whether to transmit or not is required, a random number between 0 and 1 is selected and compared to the preselected probability p. If the selected random number is equal to p or exceeds p then the subscriber unit transmits over the currently determined time slot.

If the selected random number is less then p, then an average number AVE_SS of skipped transmittals over a time period T is computed. If AVE_SS exceeds a preselected allowable constant number X of skipped

transmittals, the subscriber unit transmits over the current time slot. Otherwise, transmittal over the time slot is skipped.

Reference is now made to Fig. 87 which is a simplified pictorial illustration of a communication system constructed and operative in accordance with a preferred embodiment of the present invention, in which proximate subscriber units can communicate without base station intervention. The system of Fig. 87 comprises a base station 10100 and a plurality of subscriber units, depicted in Fig. 87 as four subscriber units 10110, 10120, 10130, and 10140. It is appreciated that the system of Fig. 87 may comprise any number of subscriber units. It is also appreciated that functions generally assigned to the base station may be otherwise assigned.

Preferably, the subscriber units 10110, 10120, 10130, and 10140 are equipped with the ability to identify their location at any given time. One example of such a system is a GPS system 10142, whereby the subscriber units are in communication with a global positioning satellite system. Other methods of identifying location, known in the prior art, may also be used.

In Fig. 87 subscriber units 10110 and 10120 are shown in communication with each other via the base station 10100. The base station mediated communication between subscriber units 10110 and 10120 may be established and maintained by any of a number of means, including means which are well-known in the prior art of mobile radio and cellular telephone systems.

In Fig. 87 subscriber units 10130 and 10140 are shown as being located proximate to one another, and are shown in direct communication with each other without the intervention of the base station 10100.

The operation of the system of Fig. 87 is now briefly described. Reference is now additionally made to

Fig. 88A, which is a simplified flowchart illustration of a preferred method of establishing and maintaining a talk around link between two subscriber units of the system of Fig. 87.

A first subscriber unit, 10130, requests a conversation with a second subscriber unit 10140 (step 10144). A decision is then made, typically at the base station 10100, as to whether to establish a mediated communication channel via the base station 10100 between the first subscriber unit 10130 and the second subscriber unit 10140, or whether to establish a direct communication channel bypassing the base station 10100 (step 10146). A direct communication channel is also termed herein a "talk around channel" or a "talk around link", while communication using such a channel is called "talk around".

In one preferred implementation of the present invention, the base station 10100 is operative to decide whether a requested conversation between a given pair of subscriber units is to be established via a mediated channel or via a direct channel as described above. Alternatively, the subscriber units may make this decision between themselves, as described below with reference to Fig. 88B.

Typically, the decision on whether to establish mediated or direct communication is based, at least partly, on information identifying the location of each subscriber unit and on a predetermined criterion, such as, for example, a criterion of proximity. The position of each subscriber unit may be known based on position information supplied by a GPS system associated with the subscriber unit; alternatively, the position information may be derived from other means. Alternatively to the use of position information, adjacency may be determined based on the signal quality of direct communication between the subscriber units. The signal quality may be

measured by any appropriate measure of signal quality, as, for example, signal strength or signal to noise ratio.

Establishing a mediated communication channel via the base station uses more air resources than establishing a direct communication channel between the subscriber units. Preferably, the base station 10100 decides to establish a mediated communication channel whenever sufficient air resources are available, and decides to establish a direct communication channel only when sufficient air resources for a mediated communication channel are not available.

Optionally, a plurality of subscriber units may be assigned to one or more groups termed herein "talk groups". For example, if there are 10,000 subscriber units, 1,200 of the subscriber units might be divided into 30 talk groups, the talk groups having varying numbers of subscriber units as members. The remaining 8,800 subscriber units might not be members of any talk group.

In the case where there are talk groups, whenever a subscriber unit which is a member of a talk group requests a connection with another member of the same talk group, a direct communication between the two member subscriber units is preferably established even when sufficient air resources are available for a mediated connection. When a subscriber unit which is not a member of a talk group requests a connection, or when a subscriber unit which is a member of a talk group requests a connection with a subscriber unit which is not a member of the same talk group, the connection request is handled as described above, for the case where there are no talk groups, with preference being given to a mediated over a direct connection.

Based on the decision of step 10146, either a direct or a mediated connection is established (step

10148). The quality of a direct link depends on many factors, including the distance between the two subscriber units, which distance may change with time, thus causing a possible degradation in quality. Preferably, a direct channel is maintained and monitored and, if necessary, it is switched automatically to a mediated channel (step 10150).

Reference is now made to Fig. 88B, which is a simplified flowchart illustration of a preferred implementation of the method of Fig. 88A. The method of Fig. 88B preferably includes the following steps:

STEP 10155: A first subscriber requests a talk around connection with a second subscriber or, even without requesting talk around, requests a conversation with second subscriber who is another member of subscriber's own talk group. Alternatively, as described above, the base station may decide whether a regular conversation request is to be handled by talk around.

STEP 10160: The second subscriber receives the talk around request and compares the quality of the received talk around request signal to a minimum signal quality threshold. The signal quality may be measured by any appropriate measure of signal quality, as, for example, signal strength or signal to noise ratio. The minimum signal threshold may, for example, be determined based on operational characteristics of the system, on subjective factors affecting users of the system, or on other criteria. A typical signal to noise ratio threshold may be, for example, approximately 15 dB.

STEP 10170: Is the signal quality of the talk around request above the minimum signal quality threshold?

STEP 10180: Reject the talk around request. If the quality of the received signal is below the minimum threshold, the talk around request is rejected because a signal quality below the threshold is taken to

be inadequate to allow a direct conversation of acceptable quality. Typically, a mediated communication channel is then established (not shown).

STEP 10190: The second subscriber sends a talk around request acknowledgement signal to the first subscriber.

STEP 10200: The first subscriber receives the talk around request acknowledgement and compares the quality thereof to a minimum threshold. The signal quality and the minimum threshold are typically similar to those described above with reference to step 10160.

STEP 10210: Is the signal quality of the talk around request acknowledgement signal above the minimum? Check to see whether the quality of the received signal is above the minimum threshold. If the quality is below the minimum, processing of the talk around request continues at step 10180, described above.

STEP 10220: Talk around link is established. A direct link is now established between the two subscriber units.

STEP 10230: During talk around conversation, continue to monitor quality and switch to regular link via base station if quality is below minimum threshold.

Typically, the subscriber units in the system of Fig. 87 are mobile subscriber units. As the subscriber units move, the signal quality may change. Other factors, such as, for example, atmospheric conditions and the presence of large buildings or geographical features may also lead to a change in signal conditions. Typically, signal quality is monitored throughout the duration of the talk around link and, should signal quality fall below a second minimum threshold, the communication link is switched to a direct link via the base station 10100.

The second minimum threshold of step 10230 may be different in value, typically lower than the first

minimum threshold described above. In other words, signal quality is allowed to degrade somewhat during the talk around conversation without causing a switch to a mediated link.

Reference is now made to Fig. 89, which is a simplified flowchart illustration of a preferred method for implementing step 10220 of Fig. 88B. The method of Fig. 89 preferably includes the following steps:

STEP 10260: Choose one of the subscriber units to play the role of the base station during the talk around conversation. The subscriber unit so chosen is termed herein the base station subscriber unit, or BS-SU. The BS-SU may be chosen by any appropriate means, including randomly. The decision must be known to both subscriber units, so that one and only one subscriber unit becomes the BS-SU. The decision is typically made arbitrarily, and is preferably made by the base station 10100.

STEP 10270: The BS-SU reverses the use of its uplink and downlink channels during the talk around conversation. In normal, mediated operation, the uplink channel is used to transmit from the subscriber unit to the base station 10100, and the downlink channel is used to receive, at the subscriber unit, transmissions originating in the base station 10100.

The uplink and downlink channels are now reversed; that is, the BS-SU transmits on the channel normally used for downlink reception, and receives on the channel normally used for uplink transmission. The BS-SU thus mimics the behavior of the base station 10100 in a normal conversation. In this way, the subscriber unit which was not selected as the BS-SU in step 10260 may transmit and receive normally.

STEP 10280: At the conclusion of the talk around conversation, the BS-SU reverts to normal operation. In other words, the BS-SU reverts to using

its uplink and downlink channels in the normal fashion, for uplink and downlink communication respectively, and thus resumes normal subscriber unit operation.

Preferably, a link established using the method of Fig. 89 is a full-duplex direct link. Alternatively, the link may be a half-duplex direct link.

Reference is now made to Fig. 90, which is a simplified flowchart illustration of an alternative preferred method for implementing the link-establishing step of Fig. 88B. The method of Fig. 90 is particularly suitable for establishing a half-duplex direct link.

In the method of Fig. 90, both subscribers use a single channel for both talking and listening. The subscriber units, in cooperation with each other or based upon instructions received externally as, for example, instructions received from the base station 10100, choose one of the available channels, either the uplink channel or the downlink channel for use during the talk around conversation (step 10290). The choice of uplink versus downlink channel may be predetermined for all subscriber units, or may be agreed upon in a communication between the two subscriber units.

During the duration of the direct link, both subscriber units use the chosen channel for both talking and listening (step 10300). The use of the single chosen channel may be controlled by any appropriate means of controlling the use of a single channel for both talking and listening, as is well known in the art.

When the direct conversation is concluded, both of the subscriber units return to normal operation, that is, both talk on the uplink and listen on the downlink (step 10310).

Reference is now made to Figs. 91 and 92. Fig. 91 is a simplified flowchart illustration of the base station side of an alternative preferred method for establishing and maintaining a talk around link between

two subscriber units of the system of Fig. 87. Fig. 92 is a simplified flowchart illustration of the subscriber unit side of an alternative preferred method for establishing and maintaining a talk around link between two subscriber units of the system of Fig. 87. The methods of Figs. 91 and 92 typically are performed in conjunction with each other.

In the method of Fig. 91, the base station 10100 extracts information describing the situation of both subscriber units, preferably including the location of the two subscriber units, whether or not the two subscriber units are members of the same talk around group, whether or not the subscriber units are subscribed to use talk around, and whether the subscriber units are located within a microsite (step 10350). A microsite is a generally small geographical area served by a transponder, and not directly from a base station. Microsites are described in US Patent 5,408,496 to Ritz et al., and in copending Israel Application Nos. 111339, 111340, 111341, and 111342, referred to above.

A check is then made to see if both subscriber units are in the same talk group and are registered to use the service (step 10360). If not, a regular call rather than a talk around call is established between the two subscriber units (step 10370).

A check is made to see whether both subscribers are in the same microsite (step 10380). If not, the distance between the two subscriber units is checked to see whether it is greater than a predetermined maximum distance, such as, for example, several hundred meters (step 10380); or if the distance is not precisely known from the GPS 10142 or otherwise (step 10390).

In step 10400, a check is made to see whether the microsite is high powered or low powered. Typically the geographical area defined by a high powered microsite is larger than the geographical area defined by a low

powered microsite.

In step 10410, both subscribers are located in a regular microsite. The base station 10100 assigns call keys, containing all of the information necessary to establish a direct talk around communication link, typically including channel frequencies allocated and frequency hopping sequences, to both subscriber units. In the case of full duplex communication, each subscriber unit is assigned two keys, with the transmission key of the first subscriber unit being identical to the reception key of the second subscriber unit, and vice versa. In the case of half duplex communication, a single identical key is assigned to each subscriber unit, and the single key is used for both transmission and reception.

At the physical layer, involving communication channels only, the base station 10100 assigns one of the two subscriber units, the role of the BS (step 10420). Typically, the choice of which subscriber unit plays the role of the base station 10100 is made randomly, but any other appropriate means may also be used to choose the subscriber unit.

The call is monitored and maintained (step 10430). Typically, the base station 10100 checks the control channels, switching the connection from a talk around connection to a regular call if necessary based on an indication of signal quality provided by the subscriber unit, or another appropriate criterion.

At call termination, the keys are deallocated so that they may be reassigned as necessary (step 10440).

In the method of Fig. 92, a subscriber unit monitors control channels until receiving a message indicating available talk around resources (step 10445).

If no control channel is received at all (step 10450), the subscriber unit attempts to establish a talk around connection using a predetermined channel assigned

in advance (step 10460).

Otherwise, the subscriber unit checks to see whether a talk around instruction was received from the base station 10100 (step 10470). If no talk around instruction was received, a regular connection is established (step 10480).

If a talk around instruction was received, the subscriber unit receives and uses a key assignment from the base station 10100 (step 10490).

During the duration of the talk around connection, the subscriber unit checks signal quality to see whether it is sufficient according to a predetermined criterion of signal quality as, for example, any of the criteria described above with reference to step 10160. (step 10500). If signal quality is insufficient, the subscriber unit, via request to the base station 10100, deallocates the talk around keys (step 10510) and establishes a regular connection (step 10480, described above).

During the talk around connection, the subscriber unit, according to instructions received from the base station 10100, operates a half duplex connection on an uplink traffic channel only, or operates a full duplex connection in accordance with keys allocated to the subscriber unit by the base station 10100 (step 10520).

At the conclusion of the call, the talk around connection is terminated and the key or keys are deallocated (step 10540).

It is appreciated that functions assigned to the base station herein, such as deciding whether a normal conversation request is to be handled by direct or by mediated communication, may also be distributed among subscriber units or may be otherwise located as, for example, in another station distinct from the base station.

Fig. 93 is a simplified pictorial illustration pf a prior art sectorized communication system.

Reference is now made to Fig. 94, which is a simplified pictorial illustration of a sectorized communication system including a region around the foot of a base station, constructed and operative in accordance with a preferred embodiment of the present invention.

~~The system of Fig. 94 comprises a base station~~ 11100, providing service to sectors 11103, 11106, and 11110. Fig. 94 also depicts an area 11115 at the foot of the base station 11100. The area 11115, at the foot of the base station 11100, is also termed herein FBS ("foot of base station") 11115. In the embodiment of Fig. 94, the FBS 11115 is treated as a separate sector.

Reference is now additionally made to Fig. 95A, which is a simplified pictorial illustration of the base station 11100 of Fig. 94, showing a plurality of sector antennas 11120 and an FBS antenna 11125. Preferably, treating the FBS 11115 as a separate sector is accomplished by providing, in addition to the antennas 11120 which serve sectors 11103, 11106, and 11110, respectively, the FBS antenna 11125 having a pattern of coverage which strongly covers the FBS 11115 and only weakly covers the other sectors.

Preferably, the FBS antenna 11125 is an antenna providing omnidirectional coverage in the horizontal plane. Preferably, the FBS antenna 11125 is disposed at a height h1 lower than the height h2 of the other sector antennas. Typically, in order to provide a cover pattern strongly covering the FBS 11115, the omnidirectional antenna is omnidirectional only in the horizontal plane and is directed toward the ground.

Reference is now made to Figs. 95B and 95C, which are simplified pictorial illustrations of an antenna coverage pattern 11128 of a preferred

implementation of the FBS antenna 11125, Fig. 95B being a top view and Fig. 95C being a side view. A suitable FBS antenna 11125, having a coverage pattern substantially similar to antenna coverage pattern 11128, may be manufactured by methods well known in the art.

Alternatively, any another type of antenna providing appropriate coverage may be used. For example, the base station sector antenna described below with reference to Fig. 97 may be used.

Preferably, the FBS antenna 11125 is fed by a dedicated transmitter and feeds a dedicated receiver. The FBS antenna 11125 may be located directly below the sector antennas 11120, or may be located elsewhere. Preferably, the FBS antenna 11125 is located near the other sector antennas 11120, that is, within a few tens or hundreds of meters therefrom. In Fig. 95A, the FBS antenna 11125 is shown as located in a separate building from the other sector antennas 11120, the base station 11100 comprising the two buildings together. It is appreciated that the FBS antenna 11125 and the other sector antennas 11120 may also be located in the same building, as shown, for example, in Fig. 97, which is described below.

Reference is now made to Fig. 96, which is a simplified pictorial illustration of an alternative embodiment of the present invention, in which FBS 11115 is incorporated into a sector 11130, FBS 11115 itself being indicated by shading in Fig. 96. Preferably, the incorporation of the FBS 11115 into the sector 11130 is accomplished by broadcasting, preferably on a control channel of the sector 11130, a signal indicating that the sector 11130 is a preferred sector and should be chosen by subscriber units, such as subscriber unit 11140, typically subscriber units located in the FBS 11115, in preference to any other sector channels being received.

Reference is now made to Fig. 97, which is a

simplified pictorial illustration of the base station 11100 of Fig. 95A, showing a plurality of sector antennas 11120 and an optional FBS antenna 11135 to provided improved coverage for the area at the foot of the base station 11115, included in sector 11130. The optional FBS antenna 11135, if present, is preferably dedicated to the FBS 11115 without providing coverage in the remainder of sector 11130. In this case, non-conflicting channels, also known as orthogonal channels, are preferably assigned to the sector antenna 11120 assigned to sector 11130 and to the optional FBS antenna 11135.

It is appreciated that other types of antennas providing an appropriate pattern of coverage as, for example, the FBS antenna 11125 described above with reference to Figs. 95A - 95C, may be used.

For any of the above preferred embodiments, a commercially available adaptive signal canceling device may optionally be used in the sector antennas 11120 to null out reception of unwanted transmissions to antennas associated with the FBS antennas 11125 and 11135. Generally, adaptive signal canceling devices provide preferred performance but may not be preferred in practice because of their expense.

Fig. 98 illustrates a wireless communication system 12001 in which the apparatus and method of the present invention can be used. A base station 12002 establishes communications with and between a plurality of mobile or portable subscriber units 12004 and a plurality of dispatch stations 12006. The subscriber units 12004 have the ability to communicate with each other and with the dispatch station 12006. The communication functions provided preferably include telephony, dispatch, one to one communications, data communications and other communication functions. The communication links are provided between the above described components and over the PSTN.

The system of Fig. 98 preferably establishes communications over a plurality of frequency channels and in a plurality of time slots. The communications over the frequency channels are preferably broken into packets which are "hopped" across the frequency channels -- thus, a communication is transmitted over more than one frequency channel in accordance with a predetermined sequence, as described in United States Patent No. 5,408,496, which is hereby incorporated by reference. Such a system is commonly referred to as a frequency hopping system.

The communications can also be "hopped" across the plurality of time slots. In this case, different parts of a communication are transmitted in different time slots, again in accordance with a predetermined sequence. Further, the communication system 12001 is shown as a sectorized system having three sectors 12008 to 12010. The hopping sequences used in the sectors 12008 to 12010 are preferably orthogonal, as explained in United States Patent No. 5,408,496. The present invention, however, is not limited to sectorized communication systems or to frequency and/or time hopped communication systems.

When a subscriber unit 12004, a dispatch station 12006 or a base station 12002 receives a signal, it is desirable to determine the state of the communication channel that the signal was received on. In Fig. 99, the preferred steps used to determine channel state, in accordance with a one embodiment of the present invention, are illustrated. In the first step 12100, after the received signals are demodulated, communication signals from one of the plurality of time slots in the time slotted communication system of Fig. 97 are detected.

For illustrative purposes only, assume that communication signals received in the time slot consists

of thirty-eight QPSK modulated symbols, each symbol falling into one of four quadrants in a modulation plane, the quadrant being specified by two bits. For example, in Fig. 100, a QPSK modulation plane having four modulation points 12102 to 12105 in quadrants 0 to 3, respectively, is shown. For each received symbol, the two bits in the symbol determine which quadrant the symbol belongs in.

After demodulation, as part of step 12100, hard detection -- a well known process -- assigns each symbol to one of the four quadrants. This process simply determines the value of the received symbol and makes the quadrant assignment. Thus, in Fig. 100, symbol S1, whose bits are 00, is assigned to quadrant 0. Symbol S2, whose bits are 01, is assigned to quadrant 2. Symbol S3, whose bits are 11, is assigned to quadrant 3. Symbol S4, whose bits are 10, is assigned to quadrant 1. This process is performed thirty-nine times, one time for each symbol in the time slot.

Next, in step 12106, each symbol is rotated to a selected one of the four quadrants, preferably quadrant 0. This process allows uniform and simpler processing of the received communication signals in accordance with the present invention. If quadrant 0 is selected as the quadrant to which all symbols are rotated to, then the symbols are rotated according to the following:

(1) If the symbol is in quadrant 0, the symbol is not rotated;

(2) If the symbol is in quadrant 2, 90° is added to the symbol phase to rotate the symbol to quadrant 0;

(3) If the symbol is in quadrant 3, 180° is added to the symbol phase to rotate the symbol to quadrant 0; and

(4) If the symbol is in quadrant 1, 270° is added to the symbol phase to rotate the symbol to quadrant 0.

Referring to Fig. 100, this process is illustrated with respect to symbols S1 to S4. Since

symbol S1 is in quadrant 0, nothing is done and symbol S1 remains in quadrant 0. Since symbol S2 is in quadrant 2, 90° is added to rotate symbol S2 to quadrant 0. Since symbol S3 is in quadrant 3, 180° is added to rotate symbol S3 to quadrant 0. Since symbol S4 is in quadrant 1, 270° is added to rotate symbol S4 to quadrant 0. Again, this process is performed thirty-nine times, one time for each symbol in the time slot.

Once all the symbols from a time slot have been rotated to the same quadrant, in step 12108, the in-phase component of each symbol in the x-y plane, which has axes intersecting the modulation points in the modulation plane as shown in Fig. 100, is determined. Since the symbol is a complex number, this is preferably performed by taking the real component of each symbol, Re(Si), where Si are the symbols in a time slot. Then, in step 12110, the quadrature component of each symbol in the x-y plane is determined. Again, since the symbols are complex numbers, this is preferably done by taking the imaginary component of each symbol in the time slot: Imag(Si). In Fig. 100, the calculation of the in-phase component, Real(S1), and the quadrature component, Imag(S1), for one symbol, S1, is illustrated. It is understood however that this process is preferably performed on each rotated symbol in a time slot.

In step 12112, the channel state of the frequency channel during the time slot that the symbols were received on is determined from the ratio of the sum of the in-phase components to the sum of the absolute value of the quadrature components. Thus, the in-phase component from each symbol in the time slot is summed. Then, the absolute value of the quadrature component from each symbol is summed. Then the channel state is the ratio of the sum of the in-phase components to the sum of the absolute value of the quadrature components. Thus, channel state, CS, is preferably determined as follows:

$$CS = \frac{\sum_{Slot} Real(Si)}{\sum_{Slot} |Imag(Si)|}$$

where Si are the symbols in the time slot. In the equation above, the absolute value is used to maintain channel state as a positive number. Other mathematical functions can be used to perform the same task.

In the above equation, it is apparent that the higher the value of CS, the higher the quality of the channel state. Conversely, the lower the value of CS, the lower the quality of the channel state.

In accordance with the present invention, two points -- the nominal modulation point and the received signal point -- are compared to determine channel state. In accordance with this embodiment of the present invention, the calculation of the in-phase and quadrature components, therefore, provides a measure of the variance or error between the received symbol and the actual transmitted symbol as determined by hard detection. If the symbols in a time slot show a small deviation, the channel state is "good" since there was not much distortion. If, on the other hand, there is a lot of deviation in the received symbols, the channel state is "poor."

It will be appreciated that while the above describes a preferred embodiment of determining channel state by processing the symbols in a time slot, the invention has broader applications. The above described processing can be performed on a single communication signal to determine the state of the communication channel on which the signal was received, although the

averaging of the received symbols yields a better indication of the channel state. Thus, it is apparent that the processing need not be restricted to the symbols in a time slot -- more or less symbols can be used as desired. Also, the use of the previously described processing steps is not limited to frequency hopping and time hopping communication systems -- they may be used on any type of communication system. In addition, if a plurality of symbols are used, they need not be rotated to one quadrant in the modulation plane as described above; instead the processing can be done within the quadrant that the symbol belongs to and the results averaged accordingly. Further, the processing of the present invention can be used with any modulation scheme.

Referring to Fig. 101, the receiving, processing and display equipment of the subscriber unit 12004 is illustrated. The receive equipment of the base station 12002 and the dispatch station 12006 is substantially similar. The transmit circuitry, which is not important to the present invention, includes a transmitter 12200 and a gain control circuit 12202 which are controlled, in part, by a frequency synthesizer 12204. Signals are transmitted through a duplexer 12206 and an antenna 12208.

On the receive side, communication signals are received on the antenna 12208 and on a second antenna 12210. Two receivers 12212 and 12214 receive the signals from the antennas 12208 and 12210, respectively. The frequency channel of reception is programmed into the receivers 12212 and 12214 by the synthesizer 12204. The receivers 12212 and 12214 are gain and frequency controlled by a circuit 12216.

The received signals from both antennas 12208 and 12210 are preferably sent to the modem 12218. The modem 12218 converts the received signals to digital signals. The modem 12218 preferably includes a digital

signal processor, preferably an Analog Devices 2111, and a ASIC device. These devices process the signals received from both receivers 12212 and 12214 in accordance with the previously described steps.

The processing is controlled by a controller 12220. The processed signals are further processed to extract voice and other information by a voice processing package 12222 and the processed communication signals are provided to a user interface 12224 through the interface 12226. The user interface 12224 includes the usual devices found in subscriber units, including a display, speakers and microphones.

The circuitry of Fig. 101, can be used to calculate the variation or error associated with received communication signals in accordance with the previously described process. Thus, the circuitry of Fig. 101 can calculate the in-phase and quadrature components associated with the received signals.

The channel state CS that is computed from received signals by the processing circuitry of Fig. 101 can be utilized to select between the two signals simultaneously received on the antennas 12208 and 12210 and by the receivers 12212 and 12214, respectively. This reception of dual signals is commonly referred to as "diversity" reception.

Referring to Fig. 102, the steps for making the selection between the two diversity signals, which are received at the same time, are illustrated. In step 12300, the channel state for the first receive channel, CS1, which is generated by the previously described process, from the signals (or signal) received on antenna 12208, is determined. In step 12302, the channel state for the second receive channel, CS2, which is generated as previously described, from the signals (or signal) received on antenna 12210, is determined. Then in step 12304, the channel states from each receive

channel are compared:

CS1 > CS2?

If CS1 is greater than CS2, then in step 12306, channel 1 is selected. Thus, the signals from antenna 12208 are selected for processing. On the other hand, if CS1 is not greater than CS2, channel 2 is selected so that the signals from the antenna 12210 are selected for processing. As before, the processing is performed by the circuitry of Fig. 101, in particular, in the digital signal processor in the modem 12218.

The selection of signals from one of two channels for processing is preferably done every time channel states are recalculated. Thus, where the channel states for the diversity channels are calculated for each time slot, it is the signals from the time slot of one of the channels that are selected for processing. A new selection is then made for every time slot. If channel state is determined from another period of signals, by way of example only, from a single signal, then the signals from the period that is used to calculate channel state are the ones selected for processing.

The channel state CS that is computed from received signals by the processing circuitry of Fig. 101 can also be utilized to erase signals which are not received with some minimum confidence level. The confidence level is preferably determined in accordance with the channel state.

Referring to Fig. 103, the preferred steps for performing erasures of signals that are not received with some minimum confidence level are shown. In step 12350, the channel state CS of a receive signal or of a group of receive signals -- for example, the signals in a selected time slot -- is determined. This can either be the channel state of the selected one of the two receive channels or it can be the channel state of a single receive channel. Then, in step 12352, the channel state

is compared to a threshold, TH. If the comparison fails, that is, if the channel state is not better than some value represented by the threshold, it is preferred to erase the signal in step 12354. If the comparison passes, that is if the channel state is better than some value represented by the threshold, in step 12356, the signal is passed on for further processing. Whether "better" means greater or less than the threshold depends on the process used to determine channel state.

The number of signals erased preferably coincides with the number of signals used to calculate the channel state. Thus, where the channel state is calculated from signals in a time slot, it is the signals from the time slot of one of the channels that are erased in step 12354. If channel state is determined from another period of signals, then the signals from the period that is used to calculate channel state are the signals that are erased in step 12354. So for example only, if channel state were calculate based on a single received signal, then it is preferred to erase only the single signal in step 12354.

The preferred value of the threshold, th, depends on the method of calculating channel state. If channel state is calculated via the in-phase and quadrature components of the received signals in a time slot, then the threshold is preferably 3.5 and the erasure is made in step 12354 if the channel state of received signals in a time slot falls below that number. If the channel state is calculated via the phase error of the received signals in a time slot, then the threshold TH is preferably 0.085 and the erasure is made in step 12354 if the channel state of the received signals in the time slot exceeds that number.

In step 12354, the erasure is preferably made by setting a metric, which is a number associated with each received signal that represents the confidence level

that the received signal was properly received, to a predetermined value, typically the lowest value. When the circuitry of Fig. 101 processes the received signals, it preferably erases the signals with the lowest metric by setting those signals to zero.

Pseudocode

The previously described steps of determining channel state from the in-phase and quadrature components of received signals from two channels, the steps of selecting between diversity signals and the steps of erasing signals which are not received with some minimum confidence level, are further described by the following pseudocode:

Calculate Channel State

```
CS_X = 0.0                    Initialize channel state
                              variables
CS_Y = 0.0
for I: = 1 to 38 do           For each symbol in a time slot
  Begin
   U:=R(I)R*(I-1)             Perform for differential
                              detection (different detection
                              equations are used for different
                              detection schemes)
    MSD, MHD:=according to routine
                              Generate metrics, MSD and MHD,
                              for soft and hard decisions,
                              respectively; see
                              the metrics generation routine
    METO[i] = TABLE2(MSD);
                              Apply MSD to Table 2 (see below)
                              which is an example of the
                              metrics preferred for a
```

```
                        specific communication system.
                        The table generates a 3 bit
                        metric, MET0[i], where i is the
                        symbol number within the time
                        slot being processed.
                        One bit of the metric MET0[i] is
                        I and two bits are the
                        confidence level.
   MET1[i] = TABLE3(Msd);
                        Apply Msd to Table 3
                        (see below). The table
                        generates a 3 bit metric,
                        MET1[i], where i is the symbol
                        number within the time slot
                        being processed. One bit of
                        the metric MET1[i] is Q and two
                        bits are the confidence level.
   If MHD = 0, then W:=U · (1-j);
                        Rotate to quadrant 0 based on
                        hard decision
   If MHD = 1, then W:=U · (-1-j);
   If MHD = 3, then W:=U · (-1+j);
   If MHD = 2, then W:=U · (1+j);
   CS_X:=CS_X +Re(W)    Sum in-phase components of
                        rotated signal
   CS_Y:=CS_Y + |Imag(W)|
                        Sum absolute value of quadrature
                        components of rotated signal
end;
```

Perform Erasure

```
If CS_X < CS_Y   CSmin, then
                        CSmin is the threshold to which
                        CS_X/CS_Y is compared to
Begin
```

```
For I:=1 to 38 do
                              For each symbol in a time slot,
                              erase by setting confidence
                              level bits to 00
  Begin
    if MET0[i] < 4 then MET0[i]:=0
                              For I = 0 (represented by 0XX,
                              i.e. < 4,
                              where 0 is the bit and XX is the
                              confidence level associated
                              with the bit) erase by setting
                              confidence level to the minimum
                              value, 00, which causes the
                              digital signal processor to
                              erase the bit. The result is
                              MET 0[i] = 0.
    if MET0[i] > 3 then MET0[i]:=4
                              For I = 1 (represented by 1XX,
                              ie. > 3,
                              where 1 is the bit and XX is the
                              confidence level associated with
                              the bit) erase by setting
                              confidence level to the minimum
                              value, 00, which causes the
                              digital signal processor to
                              erase the bit. The result is
                              MET0[i] = 4.
    if MET1[i] < 4 then MET1[i]:=0
                              For Q = 0 (represented by 0XX,
                              ie. < 4,
                              where 0 is the bit and XX is the
                              confidence level associated with
                              the bit) erase by setting
                              confidence level to the minimum
                              value, 00, which causes the
                              digital signal processor to
```

```
                              erase the bit. The result is
                              MET 1[i] = 0.
      if MET1[i]  > 3 then MET1[i]:=4
                              For I = 1 (represented by 1XX,
                              ie. > 3,
                              where 1 is the bit and XX is the
                              confidence level associated with
                              the bit) erase by setting
                              confidence level to the minimum
                              value, 00, which causes the
                              digital signal processor to
                              erase the bit. The result is
                              MET0[i] = 4.
    end;
  end;
```

Perform diversity selection

```
A0 := CS_X0   CS_Y1      0 and 1 indicate the diversity
                         channels
A1 := CS_X1   CS_Y0

For i=1 to 38
  Set M[i] = MET1[i]:MET0[i]
                         M[i] becomes a six bit metric
                         representing I and Q and the
                         confidence levels.
                         For each i, select the
                         values from channel 0 if A0 > A1
                         and select the values from
                         channel 1 if A1 ≥ A0.
```

Metrics Generation Routine

```
                         Can be calculated via table but
                         the following generator saves
```

DSP memory

Calculate $|I| = |Re(U)|$
Calculate $|Q| = |Im(U)|$
Determine a section S in the first quadrant

Referring to Fig. 104, there are 3 sectors in each quadrant. The sector that a symbol falls in is determined here. The 3 sectors are identified by S=0, 4, and 44.

If $|Q| \leq |I| \cdot \tan 30^\circ$, then S=44
If $|I| \cdot \tan 30^\circ < |Q| \leq |I| \cdot \tan 60^\circ$, then S=0
If $|I| \cdot \tan 60^\circ < |Q|$, then S=4

Calculate $\tau = (Q^2 + I^2)^{1/2}$

Calculate rings:

R=3 $0 \leq \tau \leq 1.225 \cdot 10^{-3}$

The boundaries should be less than 1; see Fig. 104 for the generated metric decision zones, including the five possible rings.

R=2 $1.225 \cdot 10^{-3} < \tau \leq 4.9 \cdot 10^{-3}$
R=1 $4.9 \cdot 10^{-3} < \tau \leq 0.0196$

R=0 $0.0196 < \tau \leq 0.49$
R=3 $\tau > 0.49$

Calculate metrics:

MSD = S+R and MHD = 0 : $Q \geq 0$, $I \geq 0$

The metrics (MHD for hard detection and MSD for soft detection) are determined here based on the values of S and R. MHD places the signal on one of four quadrants. MSD places the signal on the

metric decision zone in Fig. 98. For example, if MHD =1 (i.e. quadrant 1) and S = 44 and R =1, then MSD =17. See zone 17 in quadrant 1 of Fig. 104.

MSD = [(12-S) mod 48] + R and MHD = 1, mod indicates modular arithmetic
for Q ≥ 0, I<0

MSD = [(24+S) mod 48] + R and MHD =3,
for Q<0, I<0

MSD = [(36-S) mod 48] + R and MHD =2,
for Q<0, I ≥ 0

TABLE 2-LSB Metrics

IN	OUT	IN	OUT
0	3	24	7
1	2	25	6
2	1	26	5
3	0	27	4
4	2	28	6
5	1	29	5
6	0	30	4
7	0	31	4
8	6	32	2
9	5	33	1
10	4	34	0
11	4	35	0
12	7	36	3
13	6	37	2
14	5	38	1
15	4	39	0
16	7	40	3
17	6	41	2
18	5	42	1

19	4	43	0
20	7	44	3
21	6	45	2
22	5	46	1
23	4	47	0

TABLE 3 - MSB Metrics

IN	OUT	IN	OUT
0	3	24	7
1	2	25	6
2	1	26	5
3	0	27	4
4	3	28	7
5	2	29	6
6	1	30	5
7	0	31	4
8	3	32	7
9	2	33	6
10	1	34	5
11	0	35	4
12	3	36	7
13	2	37	6
14	1	38	5
15	0	39	4
16	2	40	6
17	1	41	5
18	0	42	4
19	0	43	4
20	6	44	2
21	5	45	1
22	4	46	0
23	4	47	0

An example will now be given that illustrates the processing of one symbol and the effect of an erasure on that symbol. Suppose that the first symbol in a time slot, S1, is received and that, when applied to the metric generation routine, yields S = 44, R = 1 and MHD =1. Applying these facts to the MSD equation:

MSD = [(12 - S) mod 48] + R = [(12 - 44) mod 48] + 1 = 17

Thus, the symbol S1 falls in zone 17 in quadrant 1 in Fig. 104.

Next MET0[1] and MET1[1] are generated by applying MSD to TABLE 2 and TABLE 3, respectively:

MET0[1] = 6 = 110

MET1[1] = 1 = 001

The first bit of MET0[1] specifies I = 1 and the next two bits, 10, specify a confidence level (00 is the lowest level, 11 is the highest). The first bit of MET1[1] specifies Q = 0 and the next two bits specify a confidence level.

Erasures are made, if at all, once all of the symbols in the time slot have been processed. If the channel state falls below a threshold, CS min, then the confidence bits for all symbols in a time slot, Si, are set to "null values", which are of low level of confidence, like 00. The processing then changes the values of the I and the Q bits to null metrics which do not contribute to an erroneous decision at the error correcting process. Essentially, the erasure means that the I and Q information content from the time slot is not used. Thus, if an erasure needs to be made, the following occurs:

MET0[1] = 4 = 100

MET1[1] = 0 = 000

This is repeated for all symbols in the time slot.

It is understood that changes may be made in

the above description without departing from the scope of the invention. It is accordingly intended that all matter contained in the above description and in the drawings be interpreted as illustrative rather than limiting. It is appreciated that the particular embodiment described in the Appendices is intended only to provide an extremely detailed disclosure of the present invention and is not intended to be limiting.

It is appreciated that the software components of the present invention may, if desired, be implemented in ROM (read-only memory) form. The software components may, generally, be implemented in hardware, if desired, using conventional techniques.

It is appreciated that various features of the invention which are, for clarity, described in the contexts of separate embodiments may also be provided in combination in a single embodiment. Conversely, various features of the invention which are, for brevity, described in the context of a single embodiment may also be provided separately or in any suitable subcombination.

It will be appreciated by persons skilled in the art that the present invention is not limited to what has been particularly shown and described hereinabove. Rather, the scope of the present invention is defined only by the claims that follow:

23097ape.ndi DN-23097PCTP 12OCT95

APPENDIX A

1. SU Automatic Gain Control (AGC) Method

1.1 General

~~The received signal level changes in time because of~~ various reasons. The AGC provides an automatic gain control to the receiver amplifiers in order to maintain a constant signal level.

There are two identical and distinct AGC methods, one for each diversity channel.

Each AGC method processes the received samples taken at each active slot.

The suffix K in each variable in this appendix denotes the index of the diversity channel. For the first diversity channel, K = 0, and for the second one, K = 1.

The AGC method is implemented at each active slot provided that the input flag (channel state) CS_FLAG(K) = 1. This flag designates that the channel state as evaluated in the slot processor for the same active slot exceeds a given threshold.

1.2 Method Block Diagram

Reference is now made to Fig. 64B, which is a simplified block diagram illustration of the method of Appendix A. As depicted in Fig. 64B for each of the 39 received complex samples, $\{r_{ki}\}$, the approximate absolute value is computed. Then the absolute values of each active slot

are summed up (SUM). The ratio of this sum to a fixed reference level is processed by computing the 20 log 10 (SUM/REF) and the result is used as the loop error. The AGC loop is a second order loop with a passive "lag-lead" loop filter. The loop filter output is integrated within a limited range of values (0-70 RES). The limited integrator output which denotes the desired attenuation in dB units, is fed to an "Attenuation to control voltage converter" and then through a D/A device to the voltage control attenuator (VCA).

1.3 Method Description

Fig. 64B is almost self explanatory. In the following we describe specific parts of the method of Fig. 64B which need more clarification.

- The AGC method is performed for each active slot, only when the slot is received with a CS_FLAG (K)=1 per channel.

- rj, j = 0....38 are the 39 filtered samples which represent the symbols of the received active slot.

- The initial condition (IC) of each IIR memory (Z^{-1}) is determined as follows:

  * IC of INTEG1 (INTEG1_IC) = 0
  * IC of SGCS (SGCS_IC) is the last value, resulted after the completion of INIT_AGC process (Chapter 3.3.4).

- $C1 = 2912 / 2^{15}$

  $C2 = 129 / 2^{15}$

  $RES = 468 / 2^{15}$

- Limiter operation

  if INTEG2 > 70*RES then SGCS = 70*RES and INTEG1 = 0

  else if INTEG2 < 0 then SGCS = 0 and INTEG1 = 0

  else SGCS = INTEG2.

Attenuation to Voltage Conversion

if SGCS < 2*RES then Y = SGCS/(10*RES)

else if SGCS < 5*RES then Y =(SGCS+RES)/(15*RES)

else if SGCS < 55*RES then Y = 0.4 + (SGCS-5*RES)/(250*RES)

else if SGCS < 65*RES then Y = 0.6 + (SGCS-55*RES)/(50*RES)

else Y = 0.8 + (SGCS-65*RES)/(25*RES).

HGCS = round(Y*2^{12})

V_CNTRL = A/D (HGCS) (Hardware operation)

HGCS should be kept in a suitable latch before and between iterations.

1.4 Definition of Variables

VARIABLE NAME	TYPE	MIN VALUE	MAX VALUE	CATEGORY
CS_FLAG K	INTEGER			INPUT
r_k	VECTOR/ COMPLEX	-1	+1	INPUT
RES	REAL	$468 / 2^{15}$		CONSTANT
C1	REAL	$2912 / 2^{15}$		CONSTANT
SUM	REAL	0	+1	TEMPORARY
ERR	REAL	-1	+1	TEMPORARY
INTEG1	REAL	-1	+1	STATIC
FIL_OUT	REAL	-1	+1	TEMPORARY
INTEG2	REAL	-1	+1	TEMPORARY
SGCS	REAL	0	+1	STATIC
HGCS	INTEGER	0	$2^{12} -1$	OUTPUT

2. BS AGC Method for TCHs

2.1 General

Reference is now made to Fig. 64C which is another simplified block diagram illustration of the method of Appendix A. The method of Fig. 64C is particularly useful for the TCH, and is similar to the one in the SU (sections 1,2, 1.3).

There are two identical and distinct AGC methods, one for each diversity channel. Each AGC method processes the received samples taken at each active slot.

The suffix K in each variable in this appendix denotes the index of the diversity channel. For the first

diversity channel, K=0 and for the second one, K=1.

The AGC method is implemented at each active slot provided that the input flag (channel state) CS_FLAG(K) = 1. This flag designates that the channel state as evaluated in the slot processor for the same active slot exceeds a given threshold.

2.2 Method Description

The figure of the AGC process (Fig. 64C) is also self-explanatory. In the following we describe specific parts of Fig. 64C which need more clarification.

a. The calculation of the absolute value and the summation are performed in the Slot Processor, while the other operations are performed in the Frame Processor.

b. The BS AGC loop is equivalent to N=8 subloops which operate concurrently. It is done by replacing each memory register Z^{-1} of Fig. 64B by a set of N=8 registers (Fig. 64C) such that each successive iteration uses a different register in a cyclic mode. The content of each register should be saved until the next time it will be used (i.e. each N=8 iteration).

c. The initial conditions of these memory registers are :

- The N=8 registers of INTEG1 : INTEG1_IC = 0

- The N=8 registers of SGCS : SGCS_IC = (20 + NP + 127)/70

where NP is the total noise power in the IF bandwidth (25

KHz) related to the receiver input and expressed in dBm. For
now NP = -132.

Note : The initial HGCS should be the response of the "Attenuation to Voltage Conversion" operation on SGCS_IC.

d. Coefficients:

RES = 468 / 2^{15}

for the first 80 (10*N) iterations:

C1 = 14558 / 2^{15}
C2 = 3234 / 2^{15}

for the next iterations :

C1 = 2912 / 2^{15}
C2 = 129 / 2^{15}

Reset INTEG1_IC to zero upon switching the coefficients (i.e. between the 80th and the 81st iterations).

e. The "limiter" and the "Attenuator to voltage conversion" operations are the same as were defined for the SU AGC method in section 1.3.

2.3 Definition of Variable

VARIABLE NAME	TYPE	MIN VALUE	MAX VALUE	CATEGORY
CS_FLAG K	INTEGER	0	1	INPUT
r_k	VECTOR / COMPLEX			INPUT
RES	REAL	$468 / 2^{15}$		CONSTANT
C1	REAL	$2912 / 2^{15}$	$14558/2^{15}$	STATIC
C2	REAL	$129 / 2^{15}$	$3234 /2^{15}$	STATIC
SUM	REAL	0	+1	TEMPORARY
ERR	REAL	-1	+1	TEMPORARY
INTEG1	REAL	-1	+1	STATIC
INTEG1_IC	REAL	0		CONSTANT
FIL_OUT	REAL	-1	+1	TEMPORARY
SGCS_IC	REAL	0	+1	CONSTANT
SGCS	REAL	0	+1	STATIC
INTEG2	REAL	-1	+1	TEMPORARY
HGCS	INTEGER	0	$2^{12} -1$	OUTPUT

3. BS AGC for ACH

The "Hardware Gain Control Signal" (HGCS) of the ACH receiver should be set invariably to the fixed value of HGCS which was used for the initial condition of the TCH AGC in section 2.2,c.

APPENDIX B

Initial AGC

In this task the AGC is set up according to the initially averaged received strength of the Control Channel. This ~~task should be performed separately over the 2 diversity~~ paths.

It is performed over filtered samples, over 4 consecutive slots at every iteration, using matched filter SU#4 with the following I/O parameters :

Input: 4 samples/symbol = 164 samples/slot.
Output: 4 samples/symbol = 164 samples/slot.

The Init AGC is performed by the following steps :

a: Reset the following parameters:

INTEG1_IC = 0
SGCS_IC = 0
ITER_COUNT = 0
TOTAL_TRY_COUNT = 0
$P^{-}(k) = 0$, k =0,1,2,3,4. where $P^{-}(k)$ is a 5 valued array
TDM_IND(i) = 0 , i = 0,1,2.

b. Set these parameters in the AGC method (Appendix A) and perform a single iteration using zero input (to set the initial HGCS).

c. Use the current 656 filtered samples R(0)....R(655) to perform the following operations. For each R(p), P

greater than or equal to 163, do :

W(p) = 1/164 X SUM { |R(p-j)| } , p = 163 ... 655

where SUM { } represents a summation over values for j = 0 to 163 ; and | | signifies the absolute value

d. Determine MAX_W = max of W(p) over p = 163 to 655.

Read 2 slots without processing them, in order to keep the next processed slots synchronized with the TDMA period.

e. Perform a single iteration of AGC method (Appendix A) with the following coefficients :

$C_1 = 14558 / 2^{15}$

$C_2 = 3234 / 2^{15}$ and with MAX_W instead of SUM.

f. If |err| is greater than or equal to 3dB and TOTAL_TRY_COUNT < 100 then

begin

TOTAL_TRY_COUNT = TOTAL_TRY_COUNT + 1

go to c

end

else continue

g. P^{-}(ITER_COUNT) = p-163 where p is the index of the sample

which maximizes W(p)

ITER_COUNT = ITER_COUNT + 1

h. TOTAL_TRY_COUNT = TOTAL_TRY_COUNT + 1
if ITER_COUNT < 5 then go to c, else continue.

For each diversity input, compute:

$\underline{P}_i = 1/5 \times \text{SUM} \{\underline{P}_i(k) \bmod 164\} \quad i = 0,1$

$\text{SIGMA}_i = 1/5 \times \text{SUM} \{P_i(k) \bmod 164 - \underline{P}_i\}^2 \quad i = 0,1$

where the summation is performed for k = 0 ...4, and i is the diversity input index (i = 0,1).

If $\text{SIGMA}_0 < \text{SIGMA}_1$ then

begin
$P^{\sim}(k) = \underline{P}_0(k), \quad k = 0..4$
$\underline{P} = \underline{P}_0$
end
else
begin
$P^{\sim}(k) = P^{\sim}_1(k), \quad k = 0..4$
$\underline{P} = \underline{P}_1$
end;
for k= 0 to 4 do
begin
m = [$P^{\sim}$(k) div 164] mod 3
TDM_IND (m) = TDM_IND (m) + 1
end;

TDM_IND_MAX = i which maximizes TDM_IND (i) over i = 0,1,2

Shift the local timing, as follows:

- P samples by hardware
- TDM_IND_MAX slots by software

The contents of the fillers memory of the AGC (INTEG1, SGCS and HGCS) should be kept unchanged until the regular AGC method is operated.

Appendix C-1: Service Quality Monitoring (S.Q.M.)

1. General

The algorithm is performed in the SU. Its purpose is to estimate the average receiving quality level and present it to the (human) subscriber in specific quantization levels. The algorithm should estimate the receiving quality of the TCHs, even when the SU is in IDLE mode and receives CCH only.

2. Input

2.1 CS_X', CS_Y'. These parameters are the filtered channel state components which are obtained from App.18, section 3.1 (ver. 0.6).

2.2 AVE_COR0 (server CCH from App.2, sec. 3)
AVE_COR1 (1st neighbor from App.2, sec. 4)
AVE_COR2 (2nd neighbor from App.2, sec. 4)

2.3 SECT_LOAD1, SECT_LOAD2. These parameters indicate the quantities of active calls in neighboring sectors / µSites numbers 1 and 2 respectively (6 bits each). The parameters are obtained from a label in the CCH as described in the CAI, section...

2.4 MODE_IND - the current SU operational mode (DISCON, IDLE, VOICE, etc.).

2.5 No_of_GCHANs - the number of frequency channels of the server BS (2 bits representing 5, 10, 15 or 20 channels). The parameter is obtained either from the site table (in the SUC NU memory) or from a lable in the CCH (CAI, section...).

3. Operation

3.1 The output of this algorithm in SQM_IND (SQM indication), which has the following possible 5 levels:

- SQM_IND = 0 indicates no receiving
- SQM_IND = 1 indicates bad receiving quality
- SQM_IND = 2 indicates fair receiving quality
- SQM_IND = 3 indicates good receiving quality
- SQM_IND = 4 indicates very good receiving quality

3.2 If MODE_IND = DISCON then SQM_IND = 0.

3.3 If MOD_IND = any applicative mode (Voice, Date etc.), then:

- While in this mode, the output SQM_IND is updated during the DTCH receiving intervals only and remains unchanged during the DTCH receiving pauses.
- For each period of N voice frames (N-TBD) do:

```
        A = 1
        if CS_X' > CS_Y'• K2 then A = 2 {K2 - TBD}
else    if CS_X' > CS_Y'• K3 then A = 3 {K3 - TBD}
else    if CS_X' > CS_Y'• K4 then A = 4 {K4 - TBD}
        SQM_IND = A
```

3.4 If MOD_IND = IDLE then:

```
-       Calculate SQM_IND as in 3.3
-       Z = AVE_CORO
-       Y = Max (AVE_COR1, AVE_COR2)
-       x = index of max AVE_COR1 (1 or 2)
-       W = SECT_LOADx
-       N = No_of_GCHANs (N = 0,1,2 or 3)
-       if (Y > Z•C1) and (W > N•M1) then SQM_IND = SQM_IND-1
else    if (Y > Z•C2) and (W > N•M2) then SQM_IND = SQM_IND-2
else    if (Y > Z•C3) and (W > N•M3) then SQM_IND = SQM_IND-3
        If SQM_IND < 1 then SQM_IND = 1
```

4. Output: SQM_IND (to MMI).

1. General

The algorithm performs a look-through into 3 special consecutive slots denoted SLS's (SYNC and Label Slots). One of them is received from the serving CCH and the other 2 from the 2 neighbouring CCH's. These 3 slots are located in 3 positions within each CCH frame of 216 slots (see CAI PS1.4). The 3 SLS's are indicated to the processor by an SLS_ITER indicator which obtained from the TIM_DIS module (section 3.4.1.3.4) The value of SLS_ITER indicates the type of each received SLS, as follows:

SLS_ITER=0 indicates the SLS of the server CCH

SLS_ITER-1 indicates the SLS of the CCH neighbour 1

SLS_ITER-2 indicates the SLS of the CCH neighbour 2

The algorithm produces the input signals to the DLL (4.12), AFC (4.11), Handoff and power control algorithms.

2. Flow Chart

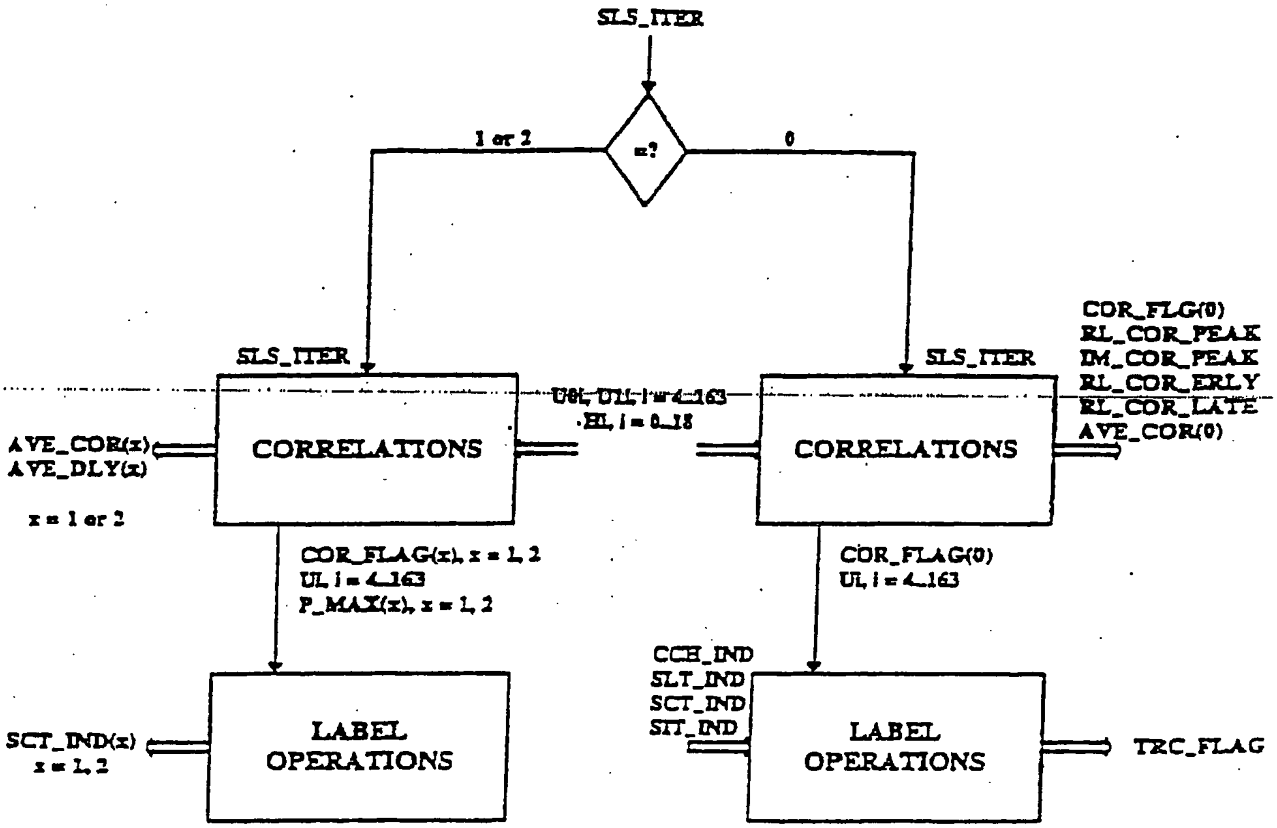


Figure A2.1: SLS Monitoring Algorithm

3. Own SLS Operations

If SLS_ITER = 0 then perform:

3.1 Input:

- {U0i}, i=4..163 {diversity channel 1}
- {U1i}, i=4..163 {diversity channel 2}
- {Hi}, i=0..18. {Hi} are the complex conjugate elements of the SYNC code which are given by:

H0	=	-1+j
H1	=	1+j
H2	=	-1-j
H3	=	1-j
H4	=	1+j
H5	=	1-j
H6	=	1+j
H7	=	-1-j
H8	=	1+j
H9	=	-1+j
H10	=	1-j
H11	=	1+j
H12	=	-1+j
H13	=	1-j
H14	=	-1-j
H15	=	-1+j
H16	=	-1-j
H17	=	-1+j
H18	=	-1-j

- CCH_IND: CCH index (0..2)
- SCT_IND: sector index (0..15)
- SLT_IND: slot index within the super frame (0..2159)
- SIT_IND: site index (0..255).

3.2 Calculate:

$$RL\_COR0 = \sum_{j=0}^{18} \mathrm{Re}\{U0(p-j*4)\} * \mathrm{Re}\{Hj\} - \mathrm{Im}\{U0(p-j*4)\} * \mathrm{Im}\{Hj\}$$

$$RL\_COR1 = \sum_{j=0}^{18} \mathrm{Re}\{U1(p-j*4)\} * \mathrm{Re}\{Hj\} - \mathrm{Im}\{U1(p-j*4)\} * \mathrm{Im}\{Hj\}$$

where $p = \hat{p}$ ($\hat{p} = 82$ is the index of the last sample of the SYNC code)

if RL_COR0 > RL_COR1 then
begin
RL_COR_PEAK = RL_COR0
{Ui} = {U0i}, i=4..163
end
else
begin
RL_COR_PEAK = RL_COR1
{Ui} = {U1i}, i=4..163
end
COR_FLG(0) = 0
if RL_COR_PEAK > Thresh then COR_FLG(0) = 1
RL_COR_EARLY = RL_COR($\hat{p}$+1) {use the above equation with Ui, i=4..163}
RL_COR_LATE = RL_COR($\hat{p}$-1) {use the above equation with Ui, i=4..163}

$$IM\_COR\_PEAK = \sum_{j=0}^{18} Re\{U(\hat{p}-j*4)\}*Im\{Hj\} + Im\{U(\hat{p}-j*4)\}*Re\{Hj\}$$

if COR_FLG(0)=1 then
AVE_COR(0) = α *AVE_COR(0) + β * RL_COR_PEAK
where AVE_COR(0)=0 in the first iteration after acquisition.

3.3 Perform Label operations as follows:

- IF COR_FLG(0) = 0 then skip section 3.3
  Else, perform:
  - Pick the 19 Ui's with the following indexes:
    $i = \hat{p}+4, \hat{p}+8, \ldots, \hat{p}+76 \quad (\hat{p}=82)$
  - For each Ui calculate Msd according to App15, sec 4.3.
    The results are 19 6-bit metrics.
- Perform Label Decoding according to App1, sec 6.
- Perform Label Verification. For each received decoded label check:
  - SCT_IND and SIT_IND are identical to those saved in memory from verification operation in DISCON mode.
  - SCT_IND mod 3 = CCH_IND (from Acq)
  - SLS_IND = [(SLT_IND+6) div 72] mod 30, where SLT_IND is the slots index within the super frame)

  Verify that this conditions are satisfied in at least K times (K=3) in a moving window of N consistent iterations (including iterations with COR_FLG(0) = 0).

If it does, then set:
TRC_FLG=1,

else, set:
TRC_FLG=0.

If the local SCT_IND is changed, then reset K and N and start the verification with the new SCT_IND.

3.4 Outputs:
- RL_COR_PEAK
- IM_COR_PEAK
- RL_COR_ERLY
- RL_COR_LATE
- COR_FLAG(0)
- AVE_COR(0)
- TRC_FLG

4. Neighbor Sector Operations

If SLS_ITER = 1 or 2 then perform:

4.1 Input:
- {U0i}, i=4..163
- {U1i}, i=4..163
- {Hi}, i=0..18. {Hi} are the complex conjugate elements of the SYNC code which are given in 3.1

Denote x = SLS_ITER = 1 or 2

4.2 Calculate:
For $p = \hat{p} - 4 \ldots \hat{p} + 4$ perform:

$$RL\_COR0(p) = \sum_{j=0}^{18} Re\{U0(p - j*4)\} * Re\{Hj\} - Im\{U0(p - j*4)\} * Im\{Hj\}$$

$$RL\_COR1(p) = \sum_{j=0}^{18} Re\{U1(p - j*4)\} * Re\{Hj\} - Im\{U1(p - j*4)\} * Im\{Hj\}$$

Determine p0 and p1 which maximize RL_COR0(p) and RL_COR1(p) respectively.

if RL_COR0(p0) > RL_COR1(p1) then
begin
    $DLY = p0 - \hat{p}$
    P_MAX(x) = p0

```
    RL_COR_PEAK = RL_COR0(p0)
    {Ui} = {U0i}, i=4..163
  end
  else
  begin
    DLY = p1 - p̂
    P_MAX(x) = p1
    RL_COR_PEAK = RL_COR1(p1)
    {Ui} = {U1i}, i=4..163
  end
  COR_FLG(x) = 0
  if RL_COR_PEAK > Thresh then COR_FLG(x) = 1
  if COR_FLG(x)=1 then
  begin
  AVE_COR(x) = α * AVE_COR(x) + β * RL_COR_PEAK
  A = γ * A + δ * RL_COR_PEAK * DLY
  B = γ * B + δ * RL_COR_PEAK
  AVE_DLY(x) = A / B
  end
```

where AVE_COR(x), A and B are zero in the first iteration after acquisition and x = SLS_ITER (1 or 2).

4.3 Perform Label operations as follows:

- IF COR_FLG(x) = 0 then skip section 4.3

Else, perform:

Pick the 19 Ui's with the following indexes:

i = P_MAX(x)+4, P_MAX(x)+8,...,P_MAX(x)+76

For each Ui, calculate Msd according to App15, sec 4.3. The results are 19 6-bit metrics.

Perform Label Decoding according to App1, sec 6.

Perform Label Verification. For each received decoded label check that SCT_IND is constant at least K times (K = 3) in a moving window of N (N = 6) consistent iterations (including iterations with COR_FLG(x) = 0).

If this condition is satisfied then set SCT_IND(x) = SCT_IND, where x = SLS_ITER (1 or 2).

4.4 Outputs:

- AVE_COR(x)
- AVE_DLY(x)
- SCT_IND(x)

where x = SLS_ITER (1 or 2)

Appendix C-3: Co-ordinated VAD Utilization (CVU)

The appendix describes the CVU process in VOICE mode. The additional operations, required for either DATA mode or composite VOICE/DATA mode are described in the "Provision for DATA" document, appendices A and B.

The VOICE CVU algorithm has 2 versions, one for the uplink and the other one for the downlink. These 2 versions are identical except for several specific points which will be noted in the following sections.

1. CVU ON

1.1 Tx Operations

1.1.1 The CVU module is obtaining voice frames with VAD="OFF" from the Vocoder (the VAD flag is a specific bit in each voice frame).

1.1.2 When the module gets a frame with VAD=ON, it should perform the following 2 subsections (concurrently):

1.1.2.1 Request for a TCH allocation, as follows:

- In the Uplink (SU):
  - Send a BACH with UTCH request and wait for a BCCH with a UTCH allocation.
  - If the SU is receiving DTCH, then perform:
    - (a) Wait a time out of T1 after transmitting the BACH and then stop receiving the DTCH at once and switch to receive the CCH.
    - (b) During the DTCH receiving pause, continue to do voice frame processing as usual, using null metrics (04) instead of the DTCH metrics which are not received.
    - (c) When receive the BCCH with the expected UTCH allocation, resume the regular receiving operations as soon as possible (start the receiving with the current active slot even if it is not the first one in its frame).
    - (d) If this BCCH was not received until T2 mseconds after the last UTCH allocation request, then perform (c) for T3 mseconds and then repeat the allocation request.
    - (e) If the local VAD turns off during this process, then perform CVU_OFF as described in section 2.1 and continue the regular receiving operations.
- In the Downlink (BS):

  Send a CVU_ON indication to the SC. The SC should allocate the FP and the SU with the same DTCH as soon as possible. If the local VAD turns off during or after this process then perform CVU_OFF as described in section 2.1.

1.1.2.2 Perform the Tx processing operations over the incoming frames from the VPP, as follows:

- If the VPP is of DVSI, then drop the first frame with VAD=ON.
- Store the next voice frames with VAD=ON in a software FIFO with depth of K frames (K=0..3, where K=0 presents no FIFO). Clear the entire FIFO, before starting this operation.
- In the Uplink (SU) perform the frame and slot processing operations (App.8 and App.15) over the first frame in the FIFO, without transmitting the result yet. Perform the slot processing operation as if the TCH key has a zero TDMA index (TDM_IND=0).
- In the Downlink (BS) perform the frame processing operation only.
- Each time the first frame is exited from the FIFO output before getting the TCH allocation, repeat the above process over the next frame and so on (keep in memory one processed frame at a time).
- As long as the TCH allocation is not received, do not send control messages through the IBOH (send them via the ACH or CCH and use the "NOP" frames with C0=C1=0, as stuffing).
- Upon receiving the TCH allocation, do:
  - In the Uplink (SU):
    (-) If the current active slot of the allocated UTCH is the first one in a frame, then start transmitting the processed frames in this slot.
    (-) Else, if it is the second one, then drop the first processed slot and start transmitting the UTCH with the second slot of the processed frame in this slot.
    (-) Else, start transmitting the processed frames in the beginning of the next UTCH frame.
    Note: If the FIFO is already full, then the processed frame will be replaced by its successive one in the beginning of the next UTCH frame.
  - In the Downlink (BS):
    Start the regular DTCH transmit process as soon as possible.

1.2 Rx Operations

1.2.1 The CVU module is delivering null voice frames (with CN=ON) to the VOCODER.

1.2.2 If the CVU module receives a message (BCCH or BACH) which includes a TCH allocation, then it performs:

- Fill the deinterleaver with all null metrics (04).
- Start performing receiving operations (frame, slot, FH, etc.) as soon as possible.
- If in the BS (receiving uplink), then the above message should include also the BACH_AMP and BACH_DLY parameters to be used as initial conditions for the AGC and DLL processes as described in App.6 and App.4 respectively.
- Send an Ack message.
- Repeat the Ack each time the module receives an IBOH message with an allocation of the same TCH.

2. CVU OFF

2.1 Tx Operations

2.1.1 The CVU module is obtaining and processing voice frames with VAD=ON.

2.1.2 When the module detects VAD=OFF, it performs:

- Denote by ON_NUM, the number of VAD_ON frames in the last talk spurt and set:

$$M = \begin{cases} M0\ \{3\} \text{ if } ON\_NUM \le ON\_NUM0 \\ M1\ \{20\} \text{ if } ON\_NUM > ON\_NUM0 \end{cases}$$

where M0, M1 and ON_NUM0 are integer constants (TBD).

- If ON_NUM > ON_NUM0, then continue to do 2.1.1 for the voice frames which remain in the FIFO and the first L {L=3} voice frames with VAD=ON (to "clean" the interleaver) and then stop transmitting.
- If ON_NUM ≤ ON_NUM0, then stop transmitting.

2.1.3 If the VAD turns ON before or at the M_th frame after the first VAD-OFF frame, then:

- Drop the first VAD-ON frame (for DVSI Vocoder only).
- Resume all transmission operations, starting with the second VAD-ON frame.
- This second VAD-ON frame should be transmitted as soon as possible after the transmission of the last VAD-ON frame of the previous spurt (part or all of the L VAD-OFF frames of section 2.1.2 which haven't been transmitted yet, should be dropped).

2.1.4 If the VAD remains "OFF" for M frames (in the FIFO output), then perform:

- Transmit a "De-alloc" command to the receiver, as follows:
  - Prepare 3 identical slots, each contains all "11" symbols.
  - Transmit them as 3 active slots in the M_th frame, without any frame processing operation.
  - Stop transmitting.
- Perform self de-allocation as follows:
  - If in the BS then notify the SC.
  - If in the SU then de-allocate the transmitter.

2.2 Rx Operations

2.2.1 The CVU module is receiving and processing voice frames (Voice and IBOH).

2.2.2 Concurrently to 2.2.1, the module searches for the "De-alloc" command, by performing the following algorithm over each received voice frame:

- Denote by M (i, j), i=0..37, j=0..2, the 6 bit metric number i, in active slot number j of the current received voice frame.

- Perform the following algorithm:

```
C=0, De-alloc_Flag=OFF
for j=0 to 2 do
for i=0 to 37 do
begin
   A = 3 LSB of M (i, j)
   B = 3 MSB of M (i, j)
   if A≤3 then A = 3-A
   if B≤3 then B = 3-B
   C = C+A+B
end; {i, j}
If C>CO {CO is a constant} then set De-alloc_Flag=ON.
```

2.2.3 If receive a frame with De-alloc_Flag=ON which is located anywhere in a "sliding window" of M1-L (17) contiguous frames (including the one with De_alloc_Flag = ON) that have at least M1-L-R-(R=2) bad CRC indications (the CRC in this case is the one of class I & half IBOH frame), then perform:

- Stop receiving the TCH
- De-allocate the host unit (BS or SU).
- Start delivering the Vocoder with null frames (any frames with CN=ON).
- If in the SU (downlink) then start receiving CCH.

If the module has missed the detection of the de-allocate command, then the de-allocation will be performed later, by the regular "Heart Beat" process.
Note: The "Heart Beat" integration period should be longer than M1 frames in order to avoid the possibility to detect "Heart Beat Failure" indication before receiving the "De_alloc_Flag" frame.

Appendix D-1: Detection, Channel State, Delay and Amplitude Estimation in the Rx_SP

1. Main Function

The module performs in general the following functions:

- Digital Matched Filtering
- Metrics generation
- Channel state
- Detects the channel delay for the DLL loop
- Estimates the average amplitude of the received slot

It operates in the Rx_SP of the SU and BS. It has several modes of operations as follows:

- Regular DTCH, BCCH or CCCH slot in the SU
- Regular UTCH or CACH slot in the BS
- UTCH slot with UTCH_SYNC (in the BS)
- Slot with SYNC (SLS and BACH). These types of slots contains a SYNC code and a DATA part. They are described elsewhere in the algorithm document. In this appendix we describe only the metrics generation of the data part of these slots (sec 4.4).

2. Timing Requirements

Start Processing: when receives the slot samples.

Finish: as soon as possible and not later than:

- In the BS: 2.20 msec (i.e. before the next active slot).
- In the SU: 6.60 msec (i.e. before the next active slot).

3. Input Variables

3.1 N complex samples from the Rx_FIFO of the first diversity branch (Rx_FIFO0). Each sample is presented by 2x12 = 24 bits. These samples are denoted by Z0(i), i=0..N-1.

3.2 N complex samples from the Rx_FIFO of the second diversity branch (Rx_FIFO1). Each sample is presented by 2•12 = 24 bits. These samples are denoted by Z1(i), i=0..N-1.

3.3 N=328 in the BS and N=164 in the SU, unless specified otherwise.

3.4 SGCS0 and SGCS1 (Software Gain Control Signals) from:

- AGC0 and AGC1 algorithms respectively if detects a regular slot (UTCH (including UTCH_SYNC), DTCH, BCCH, or CCCH)
- BACH_AMP if detects a CACH slot.

3.5 SDCS (Software Delay Control Signal) in the BS only, from:

- DLL0 algorithm if detects UTCH (including UTCH_SYNC)
- BACH_DLY if detects CACH slot.

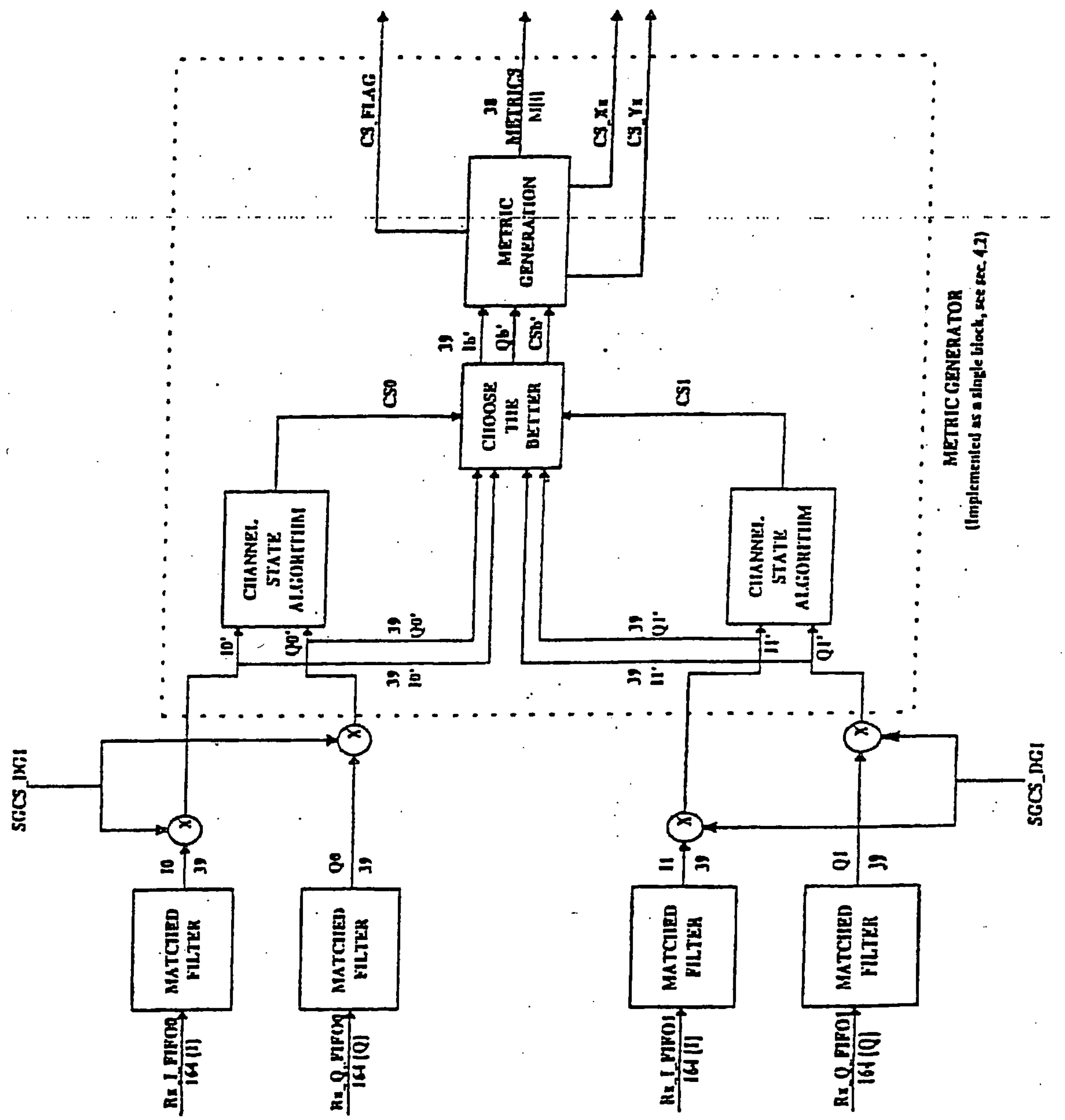


Figure A15.1: TCH Rx Slot Processing in the SU

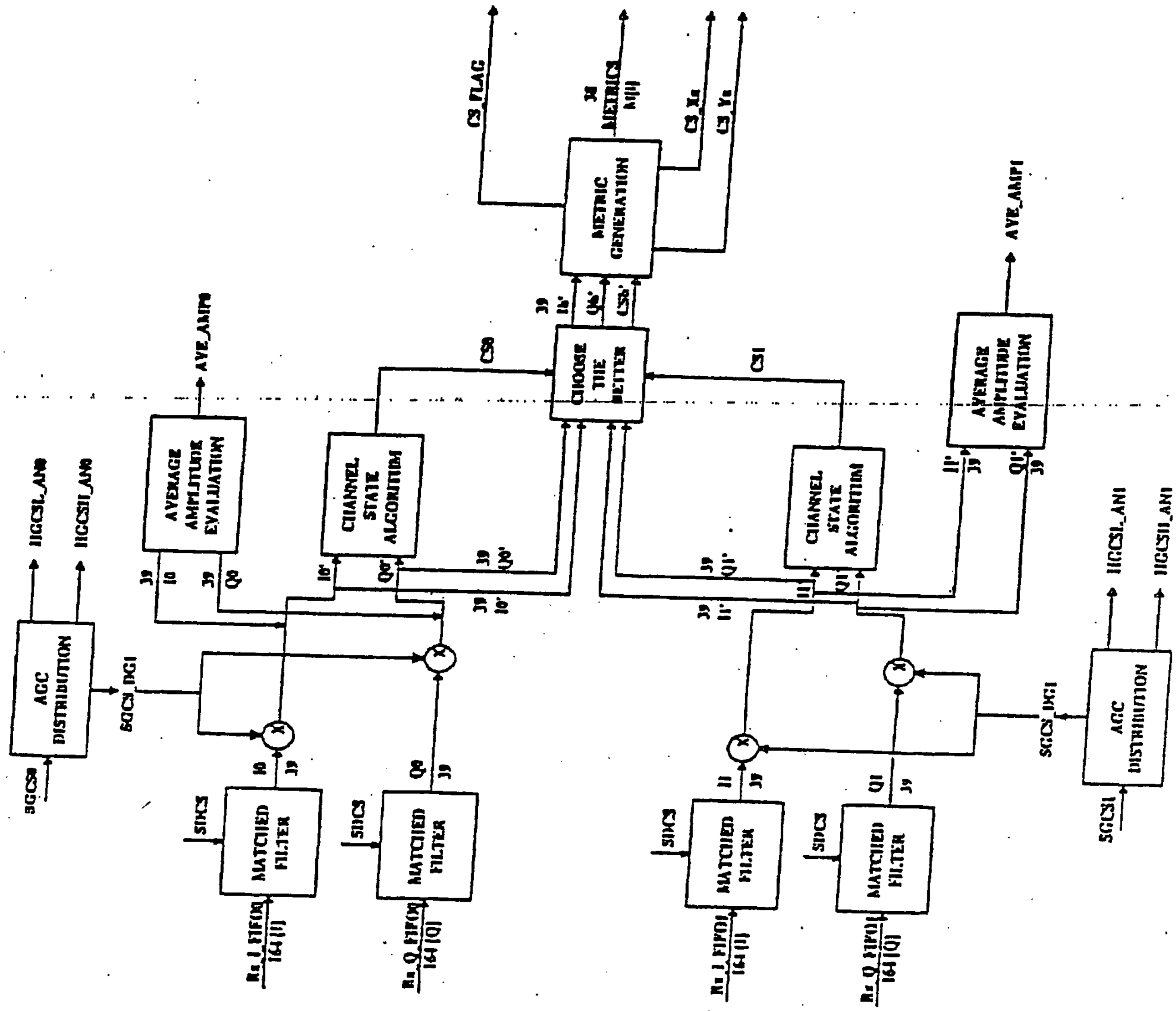


Figure A15.2: TCH Rx Slot Processing in the BS

4. Internal Operation

4.1 Digital Matched Filter

- Input:
  - Z0(i), i=0..N-1. (from first diversity input)
  - Z1(i), i=0..N-1. (from second diversity input)
- Output: 39 samples for each diversity input, denoted:
  - R0(j), j=0..38
  - R1(j), j=0..38

  except for UTCH_SYNC slot which will be described in sec. 4.3.
- Internal operation:

  Described in App. 3, Sec. 2, with R = 8.

If detect a BS slot (UTCH (including UTCH_SYNC) and CACH), then uses SDCS to shift the input as described in the appendix.

If detect a SU slot (DTCH, BCCH or CCCH) then perform the matched filtering unshifted.

4.2 Metric Generation of Regular Slots (Except SLS and BACH)

4.2.1

This section performs each diversity branch separately. Denote by Rx(j), j=0..38 the filtered samples $R_0(j)$ or $R_1(j)$ according to the diversity branch employed.

4.2.2 AGC Distribution

This operation is specified here for the BS only. For the SU it is done as part of App. 6.

Translates SGCSx by the AGC calibration table to the following 3 AGC signals:

HGCSL_ANx

HGCSH_ANx

SGCS_DGx

x indicates 0 or 1 according to the diversity branch. Set HGCSH_ANx and HGCSL_ANx to their ports (D/A converters).

4.2.3

Rx(j) := Rx(j) • SGCS_DGx, j=0..38

4.2.4 Average Amplitude Evaluation

This operation is specified here for the BS only. For the SU it is done as part of App. 6.

Calculates the average amplitude of the received slot by:

$$AVE\_AMPx = \frac{1}{39}\sum_{i=0}^{38}|Rx(i)|$$

4.2.5 Metrics Generation

Perform the following algorithm:

```
CS_X := 0.0;
CS_Y := 0.0;
for i := 1 to 38 do
   begin
         U := R(i)•R*(i-1);  {the upper asterisk indicates complex conjugate}
         Msd, Mhd:= According to sec 4.3;  {Calculate Msd and Mhd according
                                             to the algorithm in sec 4.3}
         MET0[i] := TABLE3(Msd);
         MET1[i] := TABLE4(Msd);
         if Mhd = 0 then W := U • ( 1 - j);
         if Mhd = 1 then W := U • (-1 - j);
         if Mhd = 3 then W := U • (-1 + j);
         if Mhd = 2 then W := U • ( 1 + j);
         CS_X := CS_X + Re(W)
         CS_Y := CS_Y + abs{Im(W)}
   end;
if CS_X < CS_Y • CSmin  then  {CSmin=4}
   begin
         for i := 1 to 38 do
               begin
                     if MET0[i] < 4 then MET0[i] := 0
                     if MET0[i] > 3 then MET0[i] := 4
                     if MET1[i] < 4 then MET1[i] := 0
                     if MET1[i] > 3 then MET1[i] := 4
               end;
   end;
   if CS_X > CS_Y • CSmax  then  {CSmax=9}
         begin
               for i := 1 to 38 do
                     begin
                     if MET0[i] < 4 then MET0[i] := 3
                     if MET0[i] > 3 then MET0[i] := 7;
                     if MET1[i] < 4 then MET1[i] := 3
                     if MET1[i] > 3 then MET1[i] := 7;
                     end;
         end;
         CS_FLAG := 1;
```

if CS_X < CS_Y * AGC_Thresh then {AGC_Thresh is a constant}

CS_FLAG: = 0;

where the tables are defined as follows:

- TABLE3. A fixed table which translates Msd to 3 bit metrics. The table has 48 lines of 3 bits each. It presented in table A15.2.
- TABLE4. A fixed table which translates Msd to 3 bit metrics. The table has 48 lines of 3 bits each. It is presented in table A15.3.

4.2.6 Calculate:

A0 := CS_X0 • CS_Y1

A1 := CS_X1 • CS_Y0

where the indexes 0 and 1 indicates diversity branches 0 and 1 respectively.

4.2.7 Set M[i] := MET1[i]:MET0[i], i=1..38 (i.e. M[i] is a 6-bit metric), where MET0[i] and MET1[i] are the results of 4.2.5 corresponding to:

- Diversity branch number 0 if A0 > A1
- Diversity branch number 1 if A1≥ A0

IN	OUT	IN	OUT
0	3	24	7
1	2	25	6
2	1	26	5
3	0	27	4
4	2	28	6
5	1	29	5
6	0	30	4
7	0	31	4
8	6	32	2
9	5	33	1
10	4	34	0
11	4	35	0
12	7	36	3
13	6	37	2
14	5	38	1
15	4	39	0
16	7	40	3
17	6	41	2
18	5	42	1
19	4	43	0
20	7	44	3
21	6	45	2
22	5	46	1
23	4	47	0

Table A15.1: TABLE3

IN	OUT	IN	OUT
0	3	24	7
1	2	25	6
2	1	26	5
3	0	27	4
4	3	28	7
5	2	29	6
6	1	30	5
7	0	31	4
8	3	32	7
-9	2	33	6
-10	1	34	5
-11	0	35	4
-12	3	36	7
13	2	37	6
14	1	38	5
15	0	39	4
16	2	40	6
17	1	41	5
18	0	42	4
19	0	43	4
20	6	44	2
21	5	45	1
22	4	46	0
23	4	47	0

Table A15.2: TABLE4

4.3 Msd and Mhd Metrics Generator Algorithm

1. Due to constraints in memory in the DSP, it is suggested to implement the metrics generator using quick calculation rather than a table.

2. The suggested method for calculation is as follows (see Figure A15.2):

2.1 Calculate $|I|$ and $|Q|$ and set a section S in the first quadrant, as follows:

if $|Q| \le |I| \cdot \tan 30^\circ$ then $S = 44$

if $|I| \cdot \tan 30^\circ < |Q| \le |I| \cdot \tan 60^\circ$ then $S = 0$

if $|I| \cdot \tan 60^\circ < |Q|$ then $S = 4$

Where: $I = Re(U)$, $Q = Im(U)$ and $\tan 30^\circ$, $\tan 60^\circ$ are constants (parameters).

2.3 Calculate $\gamma = \sqrt{Q^2 + I^2}$ and determine a ring R according to the boundaries of γ.

Let us mark the rings as follows:

$$\left.\begin{array}{ll} R = 3 & 0 \le \gamma \le 1.225.10^{-3} \\ R = 2 & 1.225.10^{-3} \le \gamma \le 4.9.10^{-3} \\ R = 1 & 4.9.10^{-3} \le \gamma \le 0.0196 \\ R = 0 & 0.0196 \le \gamma \le 0.49 \\ R = 3 & \gamma > 0.49 \end{array}\right\} \text{The boundaries have to be less than 1}$$

2.4 Calculate the desired metrics as follows:

$$\begin{cases} M_{SD} = S + R & : Q \ge 0,\ I \ge 0 \\ M_{HD} = 0 \end{cases}$$

$$\begin{cases} M_{SD} = [(12 - S) \bmod 48] + R & : Q \ge 0,\ I < 0 \\ M_{H} = 1 \end{cases}$$

$$\begin{cases} M_{SD} = [(24 + S) \bmod 48] + R & : Q < 0,\ I < 0 \\ M_{HD} = 3 \end{cases}$$

$$\begin{cases} M_{SD} = [(36 - S) \bmod 48] + R & : Q < 0,\ I \ge 0 \\ M_{HD} = 2 \end{cases}$$

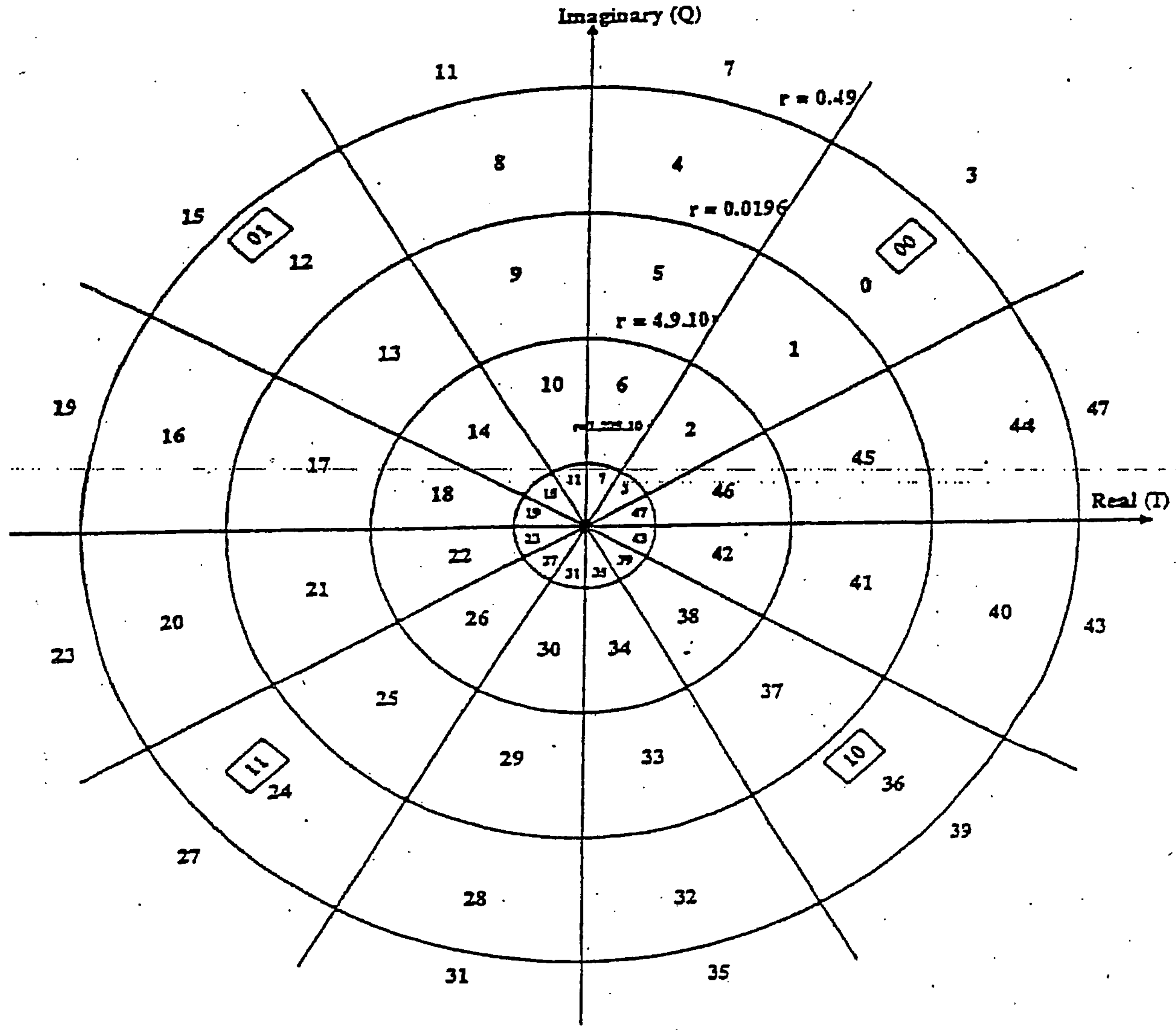


Figure A15.3: Metric Decision Zones

4.4 UTCH_SYNC Slot

4.4.1 This type of slot is received in the BS as part of the UTCH, 3 times each CCH_FRM (each 72 slots) as described in the CAI. The index x represents the diversity input (x = 0 or 1).

The next operations should be performed for both diversity inputs unless otherwise specified.

4.4.2 Generate the desired filtered samples, using the matched filter algorithm in UTCH_SYNC mode (BS#3), described in App.3, sec 8, with R=8, and the following n and d.

(a) Generate 12 filtered samples with d = SDCS-3 and n = 0..11. Denote these samples by RMx(i), i = 0..11.

(b) Generate 12 filtered samples with d = SDCS+3 and n = 0..11. Denote these samples by RPx(i), i = 0..11.

(c) Generate 39 filtered samples with d = SDCS and n = 0..38. Denote these samples by ROx(i), i = 0..38.

4.4.3 Pick the following 39 samples:

Qx(j) := ROx(j)•SGCS_DGx, j=0..38.

and calculate COR_PEAKx, using the first 12 samples as follows:

$$COR\_PEAKx = 1/12 \bullet \left|\sum_{j=0}^{11} Q_x(j) \bullet H(j)\right|$$

where H(j) is a complex number with unit absolute value ($|H(j)|=1$), representing the j-th element of the SYNC code.

- Pick the following 12 samples:
  Qx(j) := RPx(j)•SGCS_DGx, j=0..11
  and calculate COR_LATEx using the expression for COR_PEAKx.
- Pick the following 12 samples:
  Qx(j) := RMx(j)•SGCS_DGx, j=0..11
  and calculate COR_ERLYx using the expression for COR_PEAKx.
  The identifier x indicates either 0 for the first diversity branch and 1 for the second one.
- For each x, perform:
  - COR_MXCRx = max(COR_PEAKx, COR_LATEx, COR_ERLYx) where MXCR represents either PEAK, LATE or ERLY (maximum correlation).
  - For the chosen MXCR, calculate:

$$COR\_SQR\_MXCRx = 1/144 \bullet \left|\sum_{j=0}^{11} Q_x(j) \bullet H(j)\right|^2 \quad \text{(a by product of COR\_MXCRx)}$$

$$AVE\_SQR\_MXCRx = 1/12 \sum_{j=0}^{11} \left|Q_x(j)\right|^2$$

- Choose x which maximizes:

$$COR\_SQR\_MXCRx \bullet AVE\_SQR\_MXCR\bar{x}$$

over $x = 0$ and 1 where $\bar{x} = 1 - x$.

- Calculate:

if COR_SQR_MXCRx > K•AVE_SQR_MXCRx {K-Constant}, then
    D_ERR = (COR_ERLYx - COR_LATEx) / (COR_ERLYx + COR_LATEx)
else, D_ERR = 0.

4.4.4 Perform the regular metrics generation and AVE_AMP as described in section 4.2, ~~using Q0(j) and Q1(j), j=0..38~~

4.4.5 Set M[i] := 04 for i=1..11 (see sec. 4.2.7).

4.5 Slots With SYNC (SLS and BACH)

4.5.1 These types of slots contain a SYNC code and a DATA part. They are described elsewhere in the algorithm document. In this section we describe only the metrics generation of the data part of these slots.

4.5.2 Denote the number of symbols in the DATA part of the slot by N. In SLS N=20 and in BACH N=31 (each includes the reference symbol). Denote also by i=0..N-1 and by R(i) the index of the i-th sample and its value respectively.

4.5.3 Perform the following algorithm:

```
for i := 1 to N-1 do
  begin
        U := R(i)R•(i-1);  {the upper asterisk indicates complex conjugate}
        Calculate Msd according to the algorithm is sec 4.3
        MET0[i] := TABLE3(Msd);
        MET1[i] := TABLE4(Msd);
        M[i] := MET1[i]:MET0[i];
  end;
```

5. Output Variables (for regular slots only)

- M[i], i=1..38 (38 6-bit metrics)
- CS_X0 and CS_Y0 if A0 > A1 (option)
- CS_X1 and CS_Y1 if A1 >= A0 (option)
- CS_FLAG0 and CS_FLAG1
- AVE_AMP0 and AVE_AMP1 (if CS_FLAG0=1 and CS_FLAG1=1 respectively)
- D_ERR (for slots with UTCH_SYNC only. See section 4.3.3)

Appendix D-2: AGC Algorithm

1. SU Automatic Gain Control (AGC) Algorithm

1.1 General

The received signal level changes in time because of various reasons. The AGC algorithm provides an automatic gain control to the receiver amplifiers in order to maintain a constant signal level.

There are two identical and distinct AGG algorithms, one for each diversity channel. Each AGC algorithm processes the received samples taken at each active slot.

The suffix K in each variable in this appendix denotes the index of the diversity channel. For diversity channel 1, K = 0, and for diversity channel 2, K = 1.

The AGC algorithm is implemented at each active slot provided that the input flag (channel state) CS_FLAG(K) = 1. This flag designates that the channel state as evaluated in the slot processor for the same active slot exceeds a given threshold (see Appendix 15).

1.2 Algorithm Block Diagram

The block diagram of the AGC algorithm is given in figure A6.1. As depicted in the figure each of the 39 received complex samples (r_{ki}) the approximate absolute value is computed. Then the absolute values of each active slot are summed up and compared to a desired reference level (a parameter). This comparison yields a difference error, that is fed into an IIR filter that is implemented as a "Leaking Adder". The filter output (SGCS) is fed to a fixed table (AGC_TBL) which generates the control signals (HGCS) for the external voltage controlled amplifier.

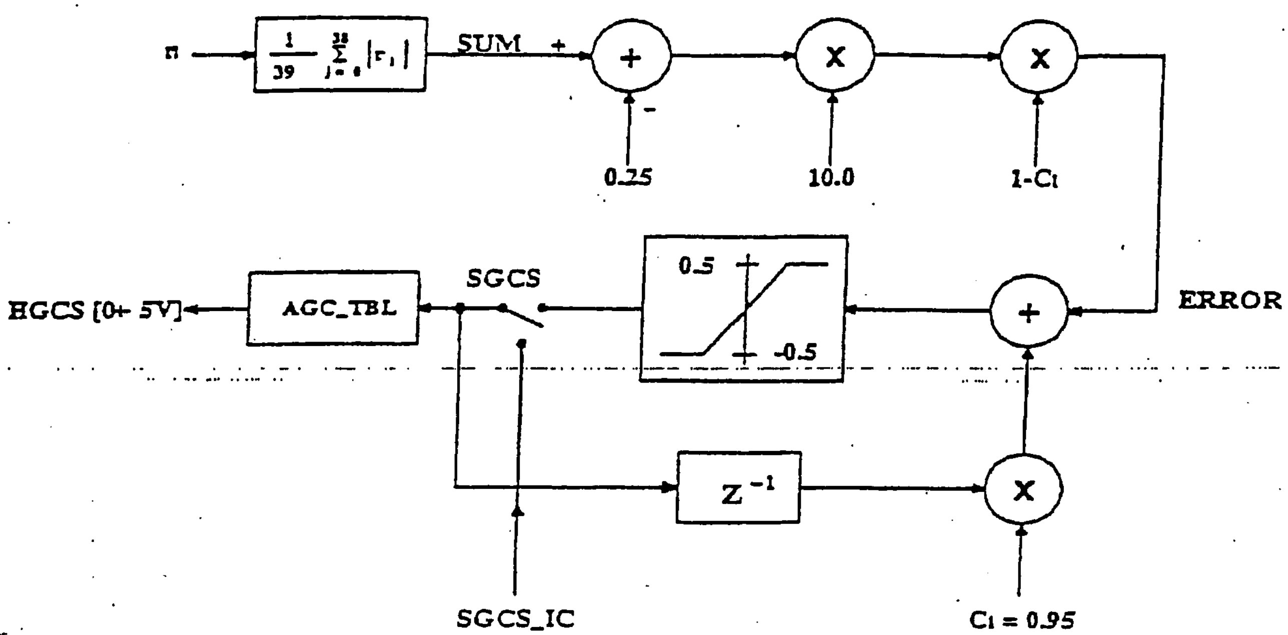


Figure A6.1: AGC Algorithm for SU

1.3 Algorithm Description

This algorithm is performed for each active slot only when the slot is received with a CS_FLAG(K) = 1 per channel. The figure (A6.1) of this algorithm is self explanatory. At each active slot the average of the absolute value of the 39 information symbols r_i is computed. Each time when a new SGCS_IC(K) is received then set it in the filter memory (Z^{-1}) and generate a new HGCS (by AGC-TBL).

1.4 Definition of Variables

VARIABLE NAME	TYPE	MIN VALUE	MAX VALUE	RESOLUTION	REPRESENTATION	CATEGORY
CS_FLAG K	INTEGER					INPUT
r_k	VECTOR COMPLEX					INPUT
~~SGCS_IC~~	~~REAL~~					~~INPUT~~
c_1	REAL	0.95	0.95			CONSTANT
REF	REAL	0.25	0.25			CONSTANT
SUM	REAL	0.0	1.0			TEMPORARY
ERROR	REAL	-0.125	0.375			TEMPORARY
FILT_MIN	REAL	-0.5	-0.5			CONSTANT
FILT_MAX	REAL	0.5	0.5			CONSTANT
SGCS	REAL	-0.5	0.5			STATIC
HGCS~~L~~	REAL	0.0	5.0			OUTPUT

Table A6.1: Variables Table

2. BS AGC Algorithm

2.1 Functions

The AGC functions in BS are similar to those described for the SU (1., this appendix). In the BS case, the calculations of the absolute value, the summation, and the AGC table are performed in the slot processor and thus are excluded from this algorithm.

The algorithm input variable AVE_AMP is proportional to the absolute value of the received signal during an active slot. This algorithm is calculated only when the CS_FLAG indicates that the received data is valid (CS_FLAG = 1). The AVE_AMP is compared to the desired received level (REF) and the difference ERROR is filtered out through "leaking integrator" type IIR filter. The filter output is applied upon completion of the iteration.

2.2 Block Diagram

The BS algorithm block diagram is given in Figure A6.2

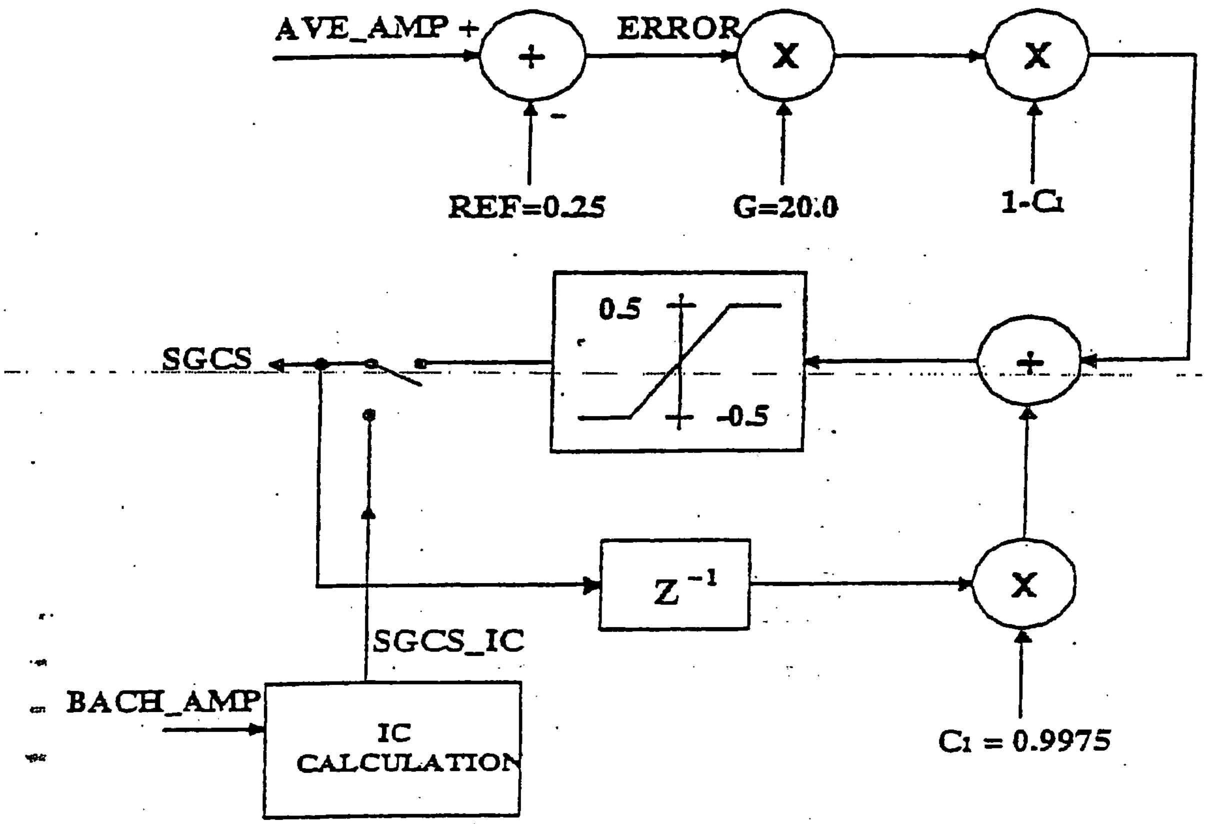


Figure A6.2: AGC Algorithm for BS

2.3 Algorithm Description

The algorithm should be performed separately for each of the 2 diversity inputs except for the IC calculation (i.e. both AGC loops use the same SGCS_IC).

These operations are performed in the TCH FP. They use the input AVE_AMP of each diversity input which are received from the SP of the appropriate TCH and yield the result 2 SGCSs (one for each diversity input) to the current SP of the same TCH.

Each received AVE_AMP is associated with a flag CS_FLAG (0 or 1).

If CS_FLAG=1 then perform the AGC loop as described in figure A6.2.

Else, skip the present iteration and deliver the result SGCS of the last iteration which was received with CS_FLAG=1.

The initial condition parameter SGCS_IC is obtained by performing the IC calculation over the input BACH_AMP, as follows:

$$SGCS\_IC = [A - 20 \cdot \log_{10}(BACH\_AMP)] \cdot AGC\_IC\_SLOPE + AGC\_IC\_DISP$$

where:

REF = 0.25
A = 17.96
AGC_IC_SLOPE = - 0.0143
AGC_IC_DISP = - 0.0714

Each time a new BACH_AMP that is relevant to the AGC operation is received, calculate SGCS_IC, set it into the filter memory (Z^{-1}) and send it as SGCS to the appropriate SPs, until the first AVE_AMP is received and the first AGC iteration is performed.

2.4 Variables

VARIABLE NAME	TYPE	MIN VALUE	MAX VALUE	RESOLUTION	REPRESENTATION	CATEGORY
CS_FLAG K	INTEGER	0	1			INPUT
AVE_AMP	REAL					INPUT
REF	REAL	0.25				CONSTANT
ERROR	REAL					TEMPORARY
BACH_AMP	REAL					INPUT
G	REAL	20				CONSTANT
A	REAL	17.96				CONSTANT
AGC_IC_SLOPE	REAL	-0.0143				CONSTANT
AGC_IC_DISP	REAL	-0.0714				CONSTANT
SGCS_IC	REAL					TEMPORARY
SGCS	REAL					OUTPUT, STATIC
C_1	REAL	0.9975				CONSTANT

Table A6.2: Variables Table

Appendix E: Discontinuous Transmitting Operation (DTO)

1. General

This algorithm is devised to stop the TCH transmission during talk pauses (in TEL, DISPATCH and DATA modes). It should be applied only when CVU is not used, since the CVU algorithm includes the DTO as part of it. The DTO algorithm is performed in the TCH transmitters and receivers both in the BS and SU. Sections 2+7 present the DTO operations in each of these modules, while in TEL or DISPATCH modes and section 8 presents the DTO operations while in DATA mode.

2. BS TCH Transmitter Operation

During a call the module can be in one of the following 2 states:

- "Silence" state, when VAD=OFF
- "Talk" state, when VAD=ON

The operation in each state and the transition between them are made as follows:

a. While in "Silence" state, the module is not transmitting (except IBOH messages as described in section 7).

b. When VAD turns ON, the module performs:
   - Switches itself to the "Talk" state.
   - Transmits a STRT_RCV marker (section 6) in the voice frame (20msec) of the first VAD_ON frame.
   - Processes and transmits the next voice frames, starting with the second VAD_ON frame (unlike the CVU, in the DTO process, the incoming voice frames should not be stored in a frame FIFO).

c. While in "Talk" state, the module is transmitting the TCH as usual.

d. When VAD turns OFF, the module performs:
   - Continues transmitting L VAD-OFF voice frames (L=3).
   - Stops transmitting for N voice frames (N=7).
   - Transmits a STOP_RCV marker (section 6) in the next frame.
   - Stops transmitting.
   - Switches itself to the "Silence" state.

   If VAD turns ON before or at the frame in which the STOP_RCV marker should be transmitted, then the module remains in "Talk" state and resumes regular transmission operation without transmitting any marker. If the VPP is of DVSI, then the module should also skip the transmission of the first VAD_ON frame.

If the module gets a control message to be sent via IBOH (section 7.2), before or at the frame in which the STOP_RCV marker should be transmitted, then the module remains in "Talk" state and resumes regular transmission operation without transmitting any marker.

3. SU TCH Receiver Operation

This module has also two possible states, "Silence" and "Talk", in accordance with the transmitter states (section 2). The operation in each state and the transition between them are done as follows:

a. While in "Silence" state, the module is performing:
   - SLS operations with DLL and AFC as in IDLE mode. (i.e. each 24 active slots).
   - AGC as usual but on the self CCH SLS slots only (if the AGC has 2 bandwidths, then use the wider one in this state).
   - Receives and processes the DTCH (as if it exists), using null metrics in slots which cannot be received due to the SLS operations.
   - Delivers the VPP with CRC flags that indicate "BAD_CRC", regardless of their actual detected value.
   - Performs IBOH operations as usual.

b. When the module detects either a STRT_RCV marker (section 6) or K "good CRC" frames in a sliding window of N frames (K=2, N=4), it performs:
   - Switches itself to the "Talk" state.
   - Delivers the VPP with the actual detected CRC flags, starting with the first frame after STRT_RCV if it was detected or with the Kth "good CRC" frame, otherwise.
   - Continue to perform IBOH operations as usual.

c. While in "Talk" state, the module is receiving the TCH as usual.

d. When the module detects either STOP_RCV marker (section 6), or M bad CRC contiguous frames (M=12), it performs:
   - Switches itself to "Silence" state.
   - Performs the operation in paragraph a of this section.

4. SU TCH Transmitter Operation

Same as the BS transmitter (section 2), except that in "Silence" state, the module should transmit the UTCH_SYNC slots (instead of 2,a) and IBOH messages (section 7).

5. BS TCH Receiver Operation

Same as the SU receiver (section 3) except that in "Silence" state the module continues the AGC and DLL operations over the received UTCH_SYNC slots (instead of SLSs). If the AGC has 2 bandwidths, then use the wider one in this state.

6. Markers

- The markers are used to provide the receivers with indications to switch states.
- There are 2 types of such markers, STRT_RCV (section 2, b) and STOP_RCV (section 2,d), which are specified as follows:
  - Air Interface format: a single voice frame (20 msec).
  - Content:
    STRT_RCV : all (114) 00 symbols
    STOP_RCV : all (114) 11 symbols
- Skip all frame processing operations (FEC, interleaving, IBOH, CRC, etc.), in the marker frame.
- If a UTCH_SYNC slot happens to be in a marker frame, then the UTCH_SYNC should be added to the marker transmission the same as in a regular voice frame.
- Each TCH receiver should try to detect these markers concurrently to the regular TCH receiving operation, as follows:

For each received TCH voice frame (114 metrics before frame processing), perform:

- Assign the metrics by M(i), i=0..113
- Calculate:

```
A = 0
for i = 0 to 113 do
begin
    X = 3 LSB of M (i)
    Y = 3 MSB of M (i)
    if X ≤ 3 then X = 3-X
    if Y ≤ 3 then Y = 3-Y
    A = A+X+Y
end;
```

The voice frame may miss part of its 114 received metrics due to either UTCH_SYNC, SLS operation or "bad" channel state (X_CS < Y_CS • K, where K is a constant). If it does, them skip the missing metrics in the above loop, and set:

Ao = U0 if it doesn't miss any metric
U1 if it misses the 12 UTCH_SYNC metrics
U2 if it misses a signal slot (38 metrics)
U3 if it misses 2 slots (76 metrics)

A1 = V0 if it doesn't miss any metric
V1 if it misses the 12 UTCH_SYNC metrics
V2 if it misses a signal slot (38 metrics)
V3 if it misses 2 slots (76 metrics)

Where U0, U1, U2, U3, V0, V1, V2 and V3 are constant (TBD).

If it misses the entire 3 slots, then stop the marker detection (i.e. ignore any possible marker in this voice frame) and continue the regular operation.
Else, perform:
if $A \leq A_0$ then STRT_RCV was received
if $A \geq A_1$ then STOP_RCV was received
else no marker was received,
where A_0 and A_1 are constant (TBD).

7. Communication with the SU During the Various States

7.1 The communication with the SU in voice mode (TEL and DISPATCH) should be done by IBOH messages.

7.2 When a TCH is in "Silence" state, then the transmitter of this TCH should transmit the following elements only (continuously in this order):
- STRT_RCV marker
- The frames of the IBOH message
- 3 tail voice frames (to "clean" the interleaver)
- STOP_RCV marker

Since the IBOH message transmission should start in an even voice frame, the STRT_RCV marker should be transmitted in the current add voice frame. The voice frames in this transmission would be those which are currently received from the VPP with VAD=OFF.

7.3 If the STRT_RCV marker is colliding with an SLS operation (in the downlink), then postpone this transmission by one IBOH frame (i.e. 2 voice frames).

7.4 If VAD turns ON during the IBOH message transmission, then the module switches itself to the "Talk" state and continues transmitting the TCH.

8. Operations in DATA mode

8.1 A TCH module (BS_FP or SU_DSP) can be set into DATA mode from either IDLE mode or from "Silence" state in TEL mode.

8.2 The first case (from IDLE mode) is not relevant to DTO.

8.3 In the second case (from "Silence" state of TEL mode), the DATA operation is limited to the same TCH which was used by the TEL operation (i.e. same source and destination as was in TEL mode). In this case, the module should stop the DTO operations and perform the DATA operations instead as described in appendices A and B of the PFD document.
Upon ending the DATA operations, it should continue the DTO as follows:

- Do not transmit any marker.
- If VAD remains OFF, then continue with the "Silence" state as usual.
- Else, the module has already been switched to the "Talk" state (see next note) and should stay in this state until situation is changes.
  Note: in DATA mode, the transmitter module should be set into, "AUTO_SWITCH_TO_VOICE_COM" = ENABLE and thus, it will automatically be switched to "Talk" state when VAD turns ON.

CLAIMS

1. A method of controlling the transmitted power in a communication system, comprising the steps of:

transmitting a signal at a first power from a first transmitter;

receiving the signal and determining the power of the signal received;

comparing the power of the signal received to a predetermined threshold and

sending the difference signal to the first transmitter;

changing the power of the signal transmitted from the first transmitter on an incremental basis.

2. The method of claim 1, wherein the first power is the maximum power.

3. A method of controlling the power of a signal transmitted from a first communication station to a second communication station where the power of the signal transmitted from the second communication station to the first communication station is known by the first communication station, comprising the steps of:

when the second communication station transmits to the first communication station, determining the power of the signal received at the first communication station;

comparing the power of the signal transmitted by the second communication station to the power received by the first communication station to determine the transmission losses;

adding the transmission losses of the signal transmitted by the second communication station to the

transmitted by the second communication station to the first communication station to the desired reception power of a signal transmitted by the first communication station to the second communication station to determine the transmitter power of the first communication station.

4. The method of claim 3, further comprising the steps of:

after a predetermined amount of time, transmitting a signal at a first power from a transmitter;

receiving the signal and determining the power of the signal received;

comparing the power of the signal received to a predetermined threshold and

sending the difference signal to the first transmitter;

changing the power of the signal transmitted from the first transmitter on an incremental basis.

5. A method of handing off communications between a communicating party and a mobile communication radio which is moving from a first sector to an adjacent sector in a communication base station having multiple sectors, comprising the steps of:

the mobile radio detecting synchronization information from the first sector and searching for synchronization information from sectors adjacent the first sector;

the mobile radio determining when the synchronization information from an adjacent sector has stronger reception than the synchronization information from the first sector and then requesting a hand off from the base station;

base station enabling a three way communication link between the mobile radio in the first sector, the radio

in the adjacent sector and the communicating party;

monitoring voice activity from the base station to the mobile radio and handing off downlink communications from the base station to the mobile radio from the first sector to the adjacent sector when voice activity is detected;

monitoring voice activity from the mobile radio to the base station and handing off uplink communications from the mobile radio base station from the first sector to the adjacent sector when voice inactivity is detected.

6. A method of controlling the gain of a time slot in a time multiplexed communication system, comprising the steps of:

averaging the amplitude values of symbols associated with a received time slot;

comparing the average amplitude with a reference signal to obtain the difference;

filtering the difference signals;

adjusting the gain of a following time slot.

7. The method of claim 2, further comprising the step of:

limiting the range of the signal being filtered.

8. The method of claim 1, wherein the reference signal is 12 dB below the maximum value of the input signal.

9. The method of claim 1, wherein the gain of the time slot immediately following the received time slot is controlled.

10. The method of claim 1, further comprising the step of adjusting the output of the filtered signals to account for nonlinearities in the amplifiers.

11. In a digital communications system including at least one base station and a multiplicity of mobile remote stations positionable at variable distances from the at least one base station,

apparatus for ensuring generally synchronous time of arrival of signals received at the base station from said mobile remote stations including:

detecting apparatus located at the base station for determining the time of arrival of signals from each mobile remote station and indicating when the time of arrival is outside a predetermined preassigned time window;

remotely controllable timing circuitry in each mobile remote station which is operative to determine the timing of transmissions therefrom to the base station; and

a timing controller, located at the base station and responsive to the output of the detecting apparatus for instructing said timing circuitry in each mobile remote station to vary the timing of its transmissions such as to cause such transmissions to be received at the base station within the predetermined preassigned time window.

12. Apparatus according to claim 11 and also comprising ranging apparatus for indicating the current distance between the base station and each of the mobile remote stations, based on time of arrival at the base station of transmissions from the mobile remote stations.

13. Apparatus according to claim 12 and wherein said ranging apparatus includes apparatus for taking into account historical variations in the timing of the transmissions of each base station.

14. Apparatus according to either of claims 12 and 13 and wherein said ranging apparatus also comprises apparatus for intermittently receiving from each of the mobile remote stations information setting forth its current transmission timing.

15. In a digital communications system including at least one transmitter transmitting multiple bit digital messages over a communications link and at least one receiver receiving said multiple bit digital messages,

apparatus associated with a receiver for dividing a received multiple bit digital message into a plurality of message segments and indicating which, if any, of such segments is received containing errors;

apparatus operative upon receipt of a retransmission of a received multiple bit digital message for replacing only such segments of the original message which were received containing errors by corresponding message segments received in the retransmission; and

apparatus operative to provide a satisfactory message receipt indication once all of the segments have been received without errors, even if no retransmission is received entirely without errors.

16. Apparatus according to claim 15 and also comprising apparatus for comparing corresponding segments of multiple transmissions of a given received multiple bit digital message to confirm that those segments identified as having been received without errors are identical for at least two transmissions.

17. A frequency-controlling signal receiving system comprising:

a frequency offset estimating modem operative

to generate an estimated frequency offset value for a signal received by the modem; and

a local oscillator operative to receive the estimated frequency offset value and to generate a frequency converter control signal which is operative to cancel the estimated frequency offset.

18. A system according to claim 16 and also comprising a frequency converter operative to receive said frequency converter control signal.

19. A system according to claim 17 or claim 18 wherein said frequency offset estimating modem comprises:

apparatus for receiving a signal with modulation and for canceling the modulation of the signal, thereby to generate a carrier wave;

an FFT (fast Fourier transform) computing module for converting the waveform of the carrier wave from a time domain to a frequency domain, thereby to generate a final spectrum function in the frequency domain; and

a processor for computing a frequency offset by finding a frequency value which maximizes the final spectrum function.

20. A system according to claim 19 wherein said FFT computing module comprises:

a converter for converting the waveform of the carrier wave from a time domain to a frequency domain, portion by portion for a plurality of portions of the waveform, thereby to generate a plurality of intermediate spectrum functions; and

spectrum function combining unit operative to combine the intermediate spectrum functions to generate the final spectrum function.

21. A system according to claim 20 wherein the spectrum function combining unit comprises a summing unit for summing the spectrum function values for each of a multiplicity of frequencies.

22. A system according to any of the claims 19 - 21 wherein the carrier wave comprises a distorted carrier wave.

23. A method for self-synchronizing a signal comprising:

detecting a synchronization code embedded in a received signal and generating timing information associated with said detected synchronization code;

providing a local timing system; and

receiving the timing information and synchronizing the local timing system in accordance with the timing information.

24. A method according to claim 23 and wherein said detecting includes:

storing a representation of the synchronization code; and

correlating between the received signal and the representation of the synchronization code.

25. A method according to claim 24 and wherein the correlating includes:

performing a sliding correlation operation within a window in which at least one element of the synchronization code is known to appear; and

finding an optimal correlation within said window,

wherein said timing information comprises an indication of a time at which said optimal correlation appears.

26. A method according to claim 24 wherein said correlating comprises:

performing a sliding correlation operation within a window in which at least N elements of the synchronization code are known to appear; and

finding M optimal correlations within said window,

wherein said timing information comprises an indication of a time at which a predetermined one from among said M optimal correlations appears.

27. A method according to claim 26 wherein K < = M optimal correlations correspond to K points in time and also comprising verifying the K optimal correlations by comparing the K points in time to a known timing pattern of the K optimal correlations, wherein the K optimal correlations form a subset of the M optimal correlations.

28. A method according to any of the claims 23 - 27 wherein said received signal comprises a synchronization code period and said synchronization code is embedded in the synchronization code period.

29. A method according to any of claims 23 - 28 wherein said received signal comprises a super frame, and

wherein a plurality of synchronization code periods are embedded in the super frame, and

wherein at least one synchronization code is embedded within each of the plurality of synchronization code periods, and

wherein a synchronization code label is associated with each synchronization code, and

wherein timing information associated with said super frame is generated and is at least in part based on the plurality of the synchronization code labels.

30. A method of maintaining a timing system in a subscriber unit in synchronization with a timing signal received from a base station in a frequency hopping multiple access communication system wherein a multiplicity of base stations communicate with a multiplicity of subscriber units over a frequency hopping multiple access communication network at a plurality of radio frequencies, the method comprising:

transmitting a timing synchronization signal at a selected slot over a control channel in said frequency hopping multiple access communication network to the subscriber unit;

receiving, over the control channel, the selected slot including the timing synchronization signal;

correlating the timing synchronization signal with a subscriber synchronization signal to provide an early correlation value and a late correlation value respectively;

generating a normalized difference between the early correlation value and the late correlation value to determine a time delay;

filtering said normalized difference with an infinite impulse response filter to provide a smooth response signal; and

providing said smooth response signal to a number controlled delay generator to generate a controlled delay correction signal.

31. A method according to claim 30 and also comprising accumulating a plurality of sequential smooth response signals smaller than a time increment at said number controlled delay generator.

32. A method according to claim 30 and also

comprising applying the controlled delay correction signal to modify the subscriber synchronization signal by a time increment substantially equal to 3 microseconds.

33. A method according to claim 31 and also comprising applying the controlled delay correction signal to modify the subscriber synchronization signal by a time increment substantially equal to 3 microseconds.

34. A method of maintaining a timing system in a base station in synchronization with a timing signal received from a subscriber unit in a frequency hopping multiple access communication system wherein a multiplicity of base stations communicate with a multiplicity of subscriber units over a frequency hopping multiple access communication network at a plurality of radio frequencies, the method comprising:

transmitting a timing synchronization signal at a selected slot over a control channel in the frequency hopping multiple access communication network to said base station;

receiving, over the control channel, the selected slot including the timing synchronization signal;

correlating the timing synchronization signal with a base station synchronization signal to provide an early correlation value and a late correlation value respectively;

generating a normalized difference between the early correlation value and the late correlation value to determine a time delay;

filtering said normalized difference with an infinite impulse response filter to provide a smooth response signal; and

providing said smooth response signal to a number controlled delay generator to generate a

controlled delay correction signal.

35. A method according to claim 34 and also comprising accumulating a plurality of sequential smooth response signals smaller than a time increment at said number controlled delay generator.

36. A method according to claim 34 and also comprising applying the controlled delay correction signal to modify the base station synchronization signal by a time increment substantially equal to 6 microseconds.

37. A method according to claim 35 and also comprising applying the controlled delay correction signal to modify the base station synchronization signal by a time increment substantially equal to 6 microseconds.

38. A delay locked loop method for controlling a local timing system of a receiver by tracking the local timing system of an IF signal generated by the receiver, the method comprising:

storing an ideal waveform of a synchronization code embedded in an output signal of an RF/IF converter in the receiver;

generating a control signal monotonically related to a difference between the timing of the IF signal generated by the receiver and the timing system of the local timing system; and

supplying the control signal to the local timing system,

wherein the step of generating a control signal comprises:

detecting first and second synchronization codes embedded in the output signal of the RF/IF

converter at times t1 and t2 respectively, where t1 is a time preceding the estimated time at which the synchronization code embedded in the output signal of the RF/IF converter is maximally correlated with said ideal waveform and t2 is a time following said estimated time;

performing a first correlation between said detected synchronization code at time t1 and said stored ideal waveform;

performing a second correlation between said detected synchronization code at time t2 and said stored ideal waveform; and

computing the difference between said first and second correlations and defining a control signal on the basis of said difference.

39. A method according to claim 38 and wherein the step of computing and defining comprises filtering the difference between said correlations.

40. A method according to claim 38 and also comprising one of extracting and inserting at least one clock period to a counter in the timing system according to the control signal.

41. A method according to claim 39 and also comprising one of extracting and inserting at least one clock period to a counter in the timing system according to the control signal.

42. A delay locked loop method according to claim 38 and wherein said local timing system is a timing system at a base station.

43. A frequency hopping multiple access communication system comprising:

a frequency hopping multiple access

communication network;

a multiplicity of base stations, at least some of which receive and transmit information at a plurality of radio frequencies over the frequency hopping multiple access communication network; and

a multiplicity of subscriber units, each receiving and transmitting information at a plurality of radio frequencies via the frequency hopping multiple access communication network, wherein each subscriber unit includes:

a frequency control unit operative to determine and to reduce inaccuracies in each separate frequency of a hopping signal received from at least one base station, to acceptable values.

44. A system according to claim 43 and wherein each subscriber unit also includes:

two separate antennas for separately receiving signals to achieve space diversity; and

two separate receivers, respectively coupled to the two separate antennas, wherein each receiver is operable to determine a quality of reception of a corresponding received signal, and said frequency control unit is selectably operable on one of said corresponding received signal having the best quality reception.

45. In a frequency hopping multiple access communication system wherein a multiplicity of base stations communicate with a multiplicity of subscriber units over a frequency hopping multiple access communication network at a plurality of radio frequencies, a method of reducing inaccuracies in frequencies of an incoming hopping signal received at a subscriber unit from at least one base station, the method comprising:

correlating the incoming signal with a

synchronization signal to provide a complex correlation signal COR_PEAK determining a frequency offset;

filtering the imaginary portion IM_COR_PEAK of said complex correlation signal to provide a smoothed frequency offset signal; and

converting the smoothed frequency offset signal to voltage for a determination of an error correction voltage signal HFCS which is applied to a voltage controlled oscillator.

46. A method according to claim 45 and wherein said filtering includes:

applying a first closed loop for integration and feedback of said IM_COR_PEAK signal to provide a preliminary filtered signal; and

applying a second closed loop including a limiter for integration and feedback of said preliminary filtered signal between a limited numerical range to provide a smoothed frequency offset signal.

47. A method according to claim 46 and wherein said limited numerical range is the numerical range between 0 and 1.

48. A method for controlling a local frequency source of a receiver which supplies a signal of a given frequency to an RF/IF converter in response to a control signal supplied to the local frequency source, the method being operative to maintain a fixed output frequency of the RF/IF converter, the method comprising:

storing an ideal waveform of a synchronization code embedded in a current output signal of the RF/IF converter;

generating a control signal monotonically related to a difference between a current output frequency of the RF/IF converter and a desired output

frequency thereof; and

supplying said control signal to said frequency source,

wherein said step of generating comprises:

detecting a synchronization code embedded in a current output signal of the RF/IF converter; and

performing a complex correlation between said detected synchronization code and said stored ideal waveform thereof and defining said control signal as the imaginary part of the result of said complex correlation.

49. A method according to claim 48 and wherein said step of generating a control signal comprises filtering the control signal.

50. A method according to claim 48 and wherein said step of generating a control signal comprises compensating for nonlinearity of operation of the local frequency source.

51. A method according to claim 49 and wherein said step of generating a control signal comprises compensating for nonlinearity of operation of the local frequency source.

52. A subscriber unit in a frequency hopping multiple access communication system comprising:

at least one antenna for accepting over-the-air RF signals;

at least one receiver, coupled to said at least one antenna, and operative to receive said RF signals and to provide an IF output of said signals;

a local frequency source, coupled to said receiver, and operative to provide a signal of a given frequency to said receiver in response to an input signal;

a memory for storing an ideal waveform of a synchronization code signal;

a processor, coupled to said local frequency source, to said at least one receiver and to said memory, and operative to determine a frequency offset between a synchronization signal embedded in said RF signals and said given frequency of said local frequency source; and

a frequency control unit operative to maintain a fixed output frequency of the receiver by employing said frequency offset to generate a control signal which controls said local frequency source.

53. Automatic gain control apparatus for use in a receiver of a slotted radio communication system, the apparatus comprising:

a sample processor receiving a plurality of samples each indicating a sampled power level of an input signal and operative to produce a processed input power signal indicating a power level of the input signal; and

error determining apparatus receiving the processed input power signal and operative to produce a power error signal.

54. Apparatus according to claim 53 and also comprising:

control apparatus receiving the power error signal and producing a gain control signal; and

a variable-gain amplifier operative to control gain of the input signal based, at least in part, on the gain control signal.

55. Apparatus according to either of claims 53 and 54 and wherein the plurality of samples comprises a plurality of samples of a current time slot.

56. Apparatus according to claim 55 and wherein the

current time slot comprises a plurality of symbols and each of the plurality of samples is associated with one of the plurality of symbols.

57. Apparatus according to either of claims 55 and 56 and wherein the variable-gain amplifier is operative to control gain of the input signal of a time slot succeeding the current time slot.

58. Apparatus according to any of claims 53 - 55 and wherein the sample processor comprises averaging apparatus operative to compute an average of the plurality of samples.

59. Apparatus according to any of claims 53 - 57 and wherein the sample processor comprises absolute value computation apparatus operative to compute an average of absolute values of the plurality of samples.

60. Apparatus according to any of claims 53 - 59 and wherein the error determining apparatus comprises logarithmic scaling apparatus operative to compute a logarithmic function of the processed input power signal.

61. Apparatus according to any of claims 53 - 60 and wherein the error determining apparatus comprises filtering apparatus.

62. Apparatus according to claim 61 and wherein the filtering apparatus comprises a lead-lag filter.

63. An automatic gain control method for use in a receiver of a slotted TDMA communication system, the method comprising:

receiving a plurality of samples each indicating a sampled power level of an input signal and

producing a processed input power signal indicating a power level of the input signal; and

receiving the processed input power signal and producing a power error signal.

64. A method according to claim 63 and also comprising:

receiving the power error signal and producing a gain control signal; and

controlling the gain of the input signal based, at least in part, on the gain control signal.

65. A method according to either of claims 63 and 64 and wherein the plurality of samples comprises a plurality of samples of a current time slot.

66. A method according to claim 65 and wherein the current time slot comprises a plurality of symbols and each of the plurality of samples is associated with one of the plurality of symbols.

67. A method according to either of claims 65 and 66 and wherein controlling comprises controlling the gain of the input signal of a time slot succeeding the current time slot.

68. A method according to any of claims 63 - 67 and wherein receiving a plurality of samples and producing a processed input power signal comprises computing an average of the plurality of samples.

69. A method according to any of claims 63 - 67 and wherein receiving a plurality of samples and producing a processed input power signal comprises computing an average of absolute values of the plurality of samples.

70. A method according to any of claims 63 - 69 and wherein receiving the processed input power signal and producing a power error signal comprises computing a logarithmic function of the processed input power signal.

71. A method according to any of claims 63 - 70 and wherein receiving the processed input power signal and producing a power error signal comprises filtering.

72. A method according to claim 71 and wherein filtering comprises filtering via a lead-lag filtering method.

73. A method for controlling the amplification of a receiver amplifier disposed upstream of a receiver modem, thereby to maintain a fixed modem input amplitude for a wide range of receiver input levels, the method comprising:

providing a current demodulator input amplitude value which is related to the amplitude of the current modem input;

converting the current demodulator input amplitude value into a logarithmic-like unit;

subtracting a desired logarithmic unit level from the converted current modem input amplitude value, thereby to provide a logarithmic-like unit difference value; and

providing a control signal to the amplifier based on said logarithmic-like unit difference value.

74. A method according to claim 73 wherein said step of providing a control signal comprises filtering said logarithmic-like unit difference value.

75. A method according to either of claim 73 or

claim 74 wherein said step of providing a control signal comprises compensating for nonlinearity of operation of the amplifier.

76. An automatic gain control method for use in a receiver of a slotted radio communication system, the method comprising:

receiving a plurality of samples each indicating a sampled power level of an input signal and producing a processed input power signal indicating a power level of the input signal; and

receiving the processed input power signal and producing a power error signal.

77. Automatic gain control apparatus for use in a receiver of a time slotted TDMA communication system, the apparatus comprising:

a sample receiver operative to receive a plurality of samples each indicating a sampled power level of an input signal and operative to produce a processed input power signal indicating a power level of the input signal; and

an input power signal receiver operative to receive the processed input power signal and produce a power error signal.

78. Apparatus for controlling the amplification of a receiver amplifier disposed upstream of a receiver modem, thereby maintaining a fixed modem input amplitude for a wide range of receiver input levels, the apparatus comprising:

an input value provider operative to provide a current demodulator input amplitude value which is related to the amplitude of the current modem input;

an input value converter operative to convert the current demodulator input amplitude value into a

logarithmic-like unit;

a subtracter operative to subtract a desired logarithmic unit level from the converted current demodulator input amplitude value, thereby providing a logarithmic-like unit difference value; and

a control-signal provider operative to provide a control signal to the amplifier based on said logarithmic-like unit difference value.

79. A method for seamlessly transferring a mobile subscriber unit, having an uplink and a downlink, from a first sector served by a first sector radio having a first antenna to a second sector served by a second sector radio having a second antenna, the method comprising:

monitoring a mobile subscriber unit in order to detect when the mobile subscriber unit passes from the first sector to the second sector; and

switching the uplink and the downlink of the mobile subscriber unit from the first antenna to the second antenna, turning off the uplink and the downlink of the first sector radio and turning on the uplink and the downlink of the second sector radio, all within a time period which is short enough to cause substantially seamless communication.

80. A method for seamlessly transferring a mobile subscriber unit, having an uplink and a downlink, from a first sector served by a first sector radio having a first antenna to a second sector served by a second sector radio having a second antenna, the method comprising:

monitoring a mobile subscriber unit in order to detect when the mobile subscriber unit passes from the first sector to the second sector;

switching the uplink of the mobile subscriber

unit from the first antenna to the second antenna, turning off the uplink of the first sector radio and turning on the uplink of the second sector radio, all while the mobile subscriber unit is not transmitting; and

switching the downlink of the mobile subscriber unit from the first antenna to the second antenna, turning off the downlink of the first sector radio and turning on the downlink of the second sector radio while the first sector radio is not transmitting.

81. A method according to claim 79 or claim 80 and also comprising adapting at least one sector radio-subscriber unit control process to the second sector radio.

82. A method according to claim 81 and wherein said at least one control process comprises time aligning.

83. A method according to claim 81 or claim 82 and wherein said at least one control process comprises power control.

84. A method according to any of claims 81 - 83 and wherein said at least one control process comprises AGC (automatic gain control) of the mobile subscriber unit.

85. A method according to any of claims 81 - 84 and wherein said at least one control process comprises sector radio AGC.

86. A method according to any of claims 81 - 85 and wherein said at least one control process comprises DLL (delay lock looping) of the mobile subscriber unit.

87. A method according to any of claims 81 - 86

wherein said at least one control process comprises sector radio DLL.

88. A method according to any of claims 81 - 86 wherein said at least one control process comprises automatic frequency control (AFC) of the mobile subscriber unit.

89. A method according to any of claims 79 - 86 wherein said switching step comprises setting up a 3-way conference between the first and second sector radios and the mobile subscriber unit.

90. Apparatus operative to seamlessly transfer a mobile subscriber unit, having an uplink and a downlink, from a first sector served by a first sector radio having a first antenna to a second sector served by a second sector radio having a second antenna, the apparatus comprising:

a mobile subscriber unit monitor operative to monitor a mobile subscriber unit in order to detect when the mobile subscriber unit passes from the first sector to the second sector; and

a link switch operative to switch the uplink and the downlink of the mobile subscriber unit from the first antenna to the second antenna, to turn off the uplink and the downlink of the first sector radio and to turn on the uplink and the downlink of the second sector radio, all within a time period which is short enough to cause substantially seamless communication.

91. Apparatus operative to seamlessly transfer a mobile subscriber unit, having an uplink and a downlink, from a first sector served by a first sector radio having a first antenna to a second sector served by a second sector radio having a second antenna, the apparatus

comprising:

a mobile subscriber unit monitor operative to monitor a mobile subscriber unit in order to detect when the mobile subscriber unit passes from the first sector to the second sector;

a link switch operative to switch the uplink of the mobile subscriber unit from the first antenna to the second antenna, to turn off the uplink of the first sector radio and to turn on the uplink of the second sector radio, all while the mobile subscriber unit is not transmitting; and

a second link switch operative to switch the downlink of the mobile subscriber unit from the first antenna to the second antenna, to turn off the downlink of the first sector radio and to turn on the downlink of the second sector radio while the first sector radio is not transmitting.

92. A power control method for use in a radio communication system comprising a first station and a second station, the method comprising:

choosing an initial transmitted power level for the first station; and

performing the following steps iteratively:

transmitting a first message from the first station to the second station;

receiving the first message at the second station;

detecting a received power level for the first message at the second station;

comparing the received power level to a predetermined value;

transmitting a second message from the second station to the first station, the second message comprising an indication of a difference between the received power level and the predetermined value;

receiving the second message at the first station; and

modifying the transmitted power level for the first station based, at least in part, on the second message.

93. A power control method according to claim 92 and wherein the initial transmitted power level is a maximum power level.

94. A power control method according to either of claims 92 or 93 and wherein the comparing comprises smoothing a signal representing the received power level.

95. A power control method according to claim 94 and wherein the smoothing is based, at least in part, on a value of the indication of difference from at least one previous iteration of the comparing.

96. A power control method according to any of claims 92 - 95 and wherein the modifying comprises:

storing the indication of the difference from the second message; and

choosing an increment for modifying the transmitted power level based, at least in part, on the indication of the difference from the second message and based, at least in part, on a stored indication of the difference from a previous iteration.

97. A power control method according to any of claims 92 - 95 and wherein the modifying comprises choosing an increment for modifying the transmitted power level based, at least in part, on a stored threshold.

98. A power control method according to claim 96 and wherein the choosing an increment is also based, at

least in part, on a stored threshold.

99. A power control method according to either of claims 97 and 98 and wherein the stored threshold is between approximately 5dB and approximately 10db.

100. A power control method according to any of claims 92 - 100 and wherein the first station comprises a subscriber unit and the second station comprises a base station.

101. A power control method for use in a radio communication system comprising a first station and a second station, the method comprising:

determining a desired received power level at the first station;

transmitting a signal from the first station to the second station, the signal comprising an indication of a nominal transmitted power level for the first station;

receiving the signal at the second station;

detecting a received power level of the signal at the second station;

comparing the received power level to the nominal transmitted power level and computing the transmission loss; and

determining the transmitted power level of the second station based, at least in part, on the desired received power level at the first station and based, at least in part, on the transmission loss.

102. A power control method according to claim 101 and wherein the signal from the first station to the second station comprises a control channel signal, and

wherein the control channel signal comprises the indication of the nominal transmitted power level.

103. A power control method according to claim 102 and wherein the comparing and computing comprises computing the difference between the received power level and the nominal transmitted power level.

104. A power control method according to either of claims 102 or 103 and wherein the determining comprises computing the sum of the desired received power level at the first station and the transmission loss.

105. A power control method according to any of claims 101 - 104 and also comprising:

determining the received power level at the first station;

comparing the received power level at the first station to a predetermined value and outputting a signal representing the difference between the received power level at the first station and the predetermined value;

transmitting a second signal from the first station to the second station, the second signal comprising an indication of the difference between the received power level at the first station and the predetermined value;

receiving the second signal at the second station; and

modifying the transmitted power level for the second station based, at least in part, on the second signal.

106. A power control method according to any of claims 101 - 105 and wherein the first station comprises a base station and the second station comprises a subscriber unit.

107. A power control method according to any of

claims 92 - 106 and wherein the detecting comprises computing a signal to noise ratio.

108. A power control method according to any of claims 92 - 106 and wherein the detecting comprises computing a bit energy to noise density ratio.

109. A power control method according to any of claims 92 - 106 and wherein the detecting comprises computing a carrier to interference ratio.

110. A power control method according to any of claims 92 - 109 and wherein the radio communication system comprises a multiple access system.

111. A power control method according to any of claims 92 - 110 and wherein the radio communication system comprises a frequency hopping system.

112. Power control apparatus for use in a radio communication system comprising a first station and a second station, the apparatus comprising:

a first station transmitter having an initial transmitted power level for the first station and operative to transmit a first message from the first station to the second station;

a second station receiver operative to receive the first message at the second station;

a power level detector operative to detect a received power level for the first message at the second station;

a power level comparator operative to compare the received power level to a predetermined value;

a second station transmitter operative to transmit a second message from the second station to the first station, the second message comprising an

indication of a difference between the received power level and the predetermined value;

a first station receiver operative to receive the second message at the first station; and

a power level controller operative to modify the transmitted power level for the first station based, at least in part, on the second message.

113. Power control apparatus for use in a radio communication system comprising a first station and a second station, the first station having a desired received power level, the apparatus comprising:

a first station transmitter operative to transmit a signal from the first station to the second station, the signal comprising an indication of a nominal transmitted power level for the first station;

a second station receiver operative to receive the signal at the second station;

a power level detector operative to measure a received power level of the signal at the second station;

a power level comparator operative to compare the received power level to the nominal transmitted power level and to compute the transmission loss; and

a power level controller operative to determine the transmitted power level for the second station based, at least in part, on the desired received power level at the first station and based, at least in part, on the transmission loss.

114. Apparatus according to either of the claims 112 or 113 and wherein the radio communication system comprises a multiple access system.

115. Apparatus according to any of claims 112 - 114 and wherein the radio communication system comprises a frequency hopping system.

116. A method for controlling the transmission power of a local transmitter according to link conditions, the method comprising:

determining link conditions by monitoring at least one characteristic of a local receiver associated with the local transmitter; and

computing a level of transmission power based on said link conditions.

117. A method according to claim 116 wherein said determining by monitoring step also comprises monitoring the transmission power of a remote transmitter which is transmitting to said local receiver and monitoring the noise floor level of a remote receiver which is receiving from said local transmitter and determining link conditions based on the local receiver characteristic, the remote transmitter transmission power, and the remote receiver noise floor level.

118. A method for controlling the transmission power of a local transmitter according to link conditions, the method comprising:

determining link conditions based on an indication of a characteristic of a remote receiver which is receiving from said local transmitter, which indication is received from the remote transmitter associated with said remote receiver; and

computing a level of transmission power based on said link conditions.

119. A method according to claim 118 and wherein said step of link condition determining also comprises:

monitoring a characteristic of a local receiver associated with the local transmitter;

initially generating an evaluation of link

conditions on the basis of said local receiver characteristic; and

improving said evaluation of link conditions upon receipt of said indication of remote receiver indication.

120. A method according to claim 119 and wherein said step of computing also comprises comparing said initially generated link conditions evaluation to said improved link conditions evaluation and taking into account the result of said comparing step when subsequently performing said initial generating step.

121. A method according to claim 120 and also comprising storing the result of said comparing step, thereby to take into account the result of the comparing step when performing more than one subsequent initial generating steps.

122. A method according to any of claims 116 - 121 and wherein said receiver characteristic comprises reception power.

123. A method according to any of claims 116 - 122 and wherein said receiver characteristic comprises the receiver's SNR (signal to noise ratio).

124. A method according to any of claims 116 - 123 and wherein said receiver characteristic comprises the receiver's SIR (signal interference ratio).

125. A method according to any of claims 116 - 124 and wherein said receiver characteristic comprises the receiver's voice frame error rate.

126. Power control apparatus for use in a radio

communication system comprising a first station and a second station, the apparatus comprising:

an initial power level chooser operative to choose an initial transmitted power level for the first station; and

an iteration controller operative to control the iterative performance of the following apparatus:

a first message transmitter operative to transmit a first message from the first station to the second station;

a first message receiver operative to receive the first message at the second station;

a received power level detector operative to detect a received power level for the first message at the second station;

a received power level comparator operative to compare the received power level to a predetermined value;

a second message transmitter operative to transmit a second message from the second station to the first station, the second message comprising an indication of a difference between the received power level and the predetermined value;

a second message receiver operative to receive the second message at the first station; and

a transmitted power level modifier operative to modify the transmitted power level for the first station based, at least in part, on the second message.

127. Power control apparatus for use in a radio communication system comprising a first station and a second station, the apparatus comprising:

a desired power level determinator operative to determine a desired received power level at the first station;

a signal transmitter operative to transmit a signal from the first station to the second station, the signal comprising an indication of a nominal transmitted power level for the first station;

a signal receiver operative to receive the signal at the second station;

a received power level detector operative to detect a received power level of the signal at the second station;

a received power level comparator operative to compare the received power level to the nominal transmitted power level and to compute the transmission loss; and

a transmitted power level determinator operative to determine the transmitted power level of the second station based, at least in part, on the desired received power level at the first station and based, at least in part, on the transmission loss.

128. A power control method for use in a radio communication system comprising a first station and a second station, the method comprising:

transmitting an initial transmitted power level message from the first station to the second station;

receiving the first message at the second station;

detecting a received power level for the first message at the second station;

comparing the received power level to a predetermined value;

transmitting a second message from the second station to the first station, the second message comprising an indication of a difference between the received power level and the predetermined value;

receiving the second message at the first station; and

modifying the transmitted power level for the first station based, at least in part, on the second message.

129. A power control method for use in a radio communication system comprising a first station and a second station, the first station having a desired received power level, the method comprising:

transmitting a signal from the first station to the second station, the signal comprising an indication of a nominal transmitted power level for the first station;

receiving the signal at the second station;

detecting operative to measure a received power level of the signal at the second station;

comparing the received power level to the nominal transmitted power level and to compute the transmission loss; and

determining the transmitted power level for the second station based, at least in part, on the desired received power level at the first station and based, at least in part, on the transmission loss.

130. Apparatus for controlling the transmission power of a local transmitter according to link conditions, the apparatus comprising:

a link condition determinator operative to determine link conditions by monitoring at least one characteristic of a local receiver associated with the local transmitter; and

a transmission power level computer operative to compute a level of transmission power based on said link conditions.

131. Apparatus for controlling the transmission power of a local transmitter according to link

conditions, the apparatus comprising:

a link condition determinator operative to determine link conditions based on an indication of a characteristic of a remote receiver which is receiving from said local transmitter, which indication is received from the remote transmitter associated with said remote receiver; and

a transmission power level computer operative to compute a level of transmission power based on said link conditions.

132. A method for time alignment of messages in a radio communication system having a first station and a second station, the method comprising:

determining a time alignment error of a message sent by the first station and received by the second station;

sending a time alignment adjustment message from the second station to the first station, the time alignment adjustment message comprising a signal indicating the time alignment error of the message sent by the first station; and

adjusting the timing of subsequent messages sent by the first station to the second station based, at least in part, on the time alignment adjustment message.

133. A method according to claim 132 and wherein the determining step comprises comparing the time alignment error to a minimum error, and

wherein the sending and adjusting are performed only if the time alignment error is greater in magnitude than the minimum error.

134. A method according to either of the claims 132 and 133 and wherein the message sent by the first station comprises a station identification, and

wherein the determining step includes comparing the station identification to a stored station identification of the first station, and

wherein the sending and adjusting steps are performed only if the station identification matches the stored station identification.

135. A method according to any of claims 132 - 134 and wherein the message sent by the first station comprises a message type identification, and

wherein the time alignment adjustment message comprises the message type identification.

136. A method according to claim 135 and wherein the adjusting step is performed only if the message type identification matches a stored message type identification corresponding to a previous message sent by the first station.

137. A method according to any of claims 132 - 136 and wherein the adjusting step comprises incrementally adjusting the timing of the subsequent messages based, at least in part, on a maximum adjustment for each of the subsequent messages.

138. A method according to claim 137 and wherein the subsequent messages comprise messages each having a message type and wherein the maximum adjustment for each one of the subsequent messages is based, at least in part, on the message type of said each one of the subsequent messages.

139. A method according to any of claims 132 - 138 and also comprising determining the distance between the first station and the second station based, at least in part, on the time alignment error.

140. A method according to any of claims 132 - 138 and also comprising:

storing time alignment errors of each of a plurality of messages; and

determining the distance between the first station and the second station based, at least in part, on the stored time alignment errors.

141. Apparatus for time alignment of messages in a radio communication system having a first station and a second station, the apparatus comprising:

a time alignment determiner operative to determine a time alignment error of a message sent by the first station and received by the second station;

a message transmitter operative to send a time alignment adjustment message from the second station to the first station, the time alignment adjustment message comprising a signal indicating the time alignment error of the message sent by the first station; and

a time alignment adjustor operative to adjust the timing of subsequent messages sent by the first station to the second station based, at least in part, on the time alignment adjustment message.

142. Apparatus according to claim 141 and wherein the time alignment determiner comprises a time alignment comparator operative to compare the time alignment error to a minimum error, and

wherein the message transmitter sends the time alignment adjustment message and the time alignment adjustor adjusts the timing only if the time alignment error is greater in magnitude than the minimum error.

143. Apparatus according to either of claims 141 and 142 and wherein the message sent by the first station

comprises a station identification, and

wherein the time alignment determiner comprises a station identification comparator operative to compare the station identification to a stored station identification of the first station, and

wherein the message transmitter sends the time alignment adjustment message and the time alignment adjustor adjusts the timing only if the station identification matches the stored station identification.

144. Apparatus according to any of claims 141 - 143 and wherein the message sent by the first station comprises a message type identification, and

wherein the time alignment adjustment message comprises the message type identification.

145. Apparatus according to claim 144 and wherein the time alignment adjustor adjusts the timing only if the message type identification matches a stored message type identification corresponding to a previous message sent by the first station.

146. Apparatus according to any of claims 141 - 145 and wherein the alignment adjustor is operative to adjust the timing of the subsequent messages incrementally based, at least in part, on a maximum adjustment for each of the subsequent messages.

147. Apparatus according to claim 146 and wherein the subsequent messages comprise messages each having a message type and wherein the maximum adjustment for each one of the subsequent messages is based, at least in part, on the message type of said each one of the subsequent messages.

148. Apparatus according to any of claims 141 -- 147

and also comprising a range determiner operative to determine the distance between the first station and the second station based, at least in part, on the time alignment error.

149. Apparatus according to any of claims 141 - 148 and also comprising:

a time alignment memory operative to store time alignment errors of each of a plurality of messages; and

a distance determiner operative to determine the distance between the first station and the second station based, at least in part, on the stored time alignment errors.

150. Apparatus for time alignment of uplink transmissions and comprising:

a plurality of time aligning subscriber units; and

a time alignment determining base station operative to determine the time at which each uplink transmission arrives at the base station from a corresponding subscriber unit, to compute the timing error of each uplink transmission by comparing the time of arrival thereof to a desired time of arrival, and to transmit each said timing error to the corresponding subscriber unit,

wherein each said time aligning subscriber unit is operative to align its timing in order to reduce its timing error.

151. A method for time alignment of uplink transmissions arriving from a plurality of subscriber units, the method comprising:

determining the time at which each uplink transmission arrives at a base station;

computing the timing error of each uplink

transmission by comparing the time of arrival thereof to a desired time of arrival;

transmitting each timing error to the subscriber unit corresponding thereto; and

at each subscriber unit, aligning timing of a subsequent uplink transmission in order to reduce its timing error.

152. A method according to claim 151 wherein said timing alignment step comprises performing a plurality of partial timing aligning substeps so as to gradually reduce the timing error.

153. A method for processing received messages in order to reduce repeat transmissions of messages, the method comprising:

transmitting a message including a plurality of sub-messages;

requesting retransmission of at least one incorrectly received sub-message; and

if at least one previously incorrectly received sub-message is correctly received by retransmission, replacing at least one previously incorrectly received sub-message with its corresponding correctly received sub-message.

154. A method according to claim 153 and wherein the transmitting step comprises transmitting via a frequency hopping multiple access (FHMA) communication system.

155. A method according to any of claims 153 - 154 and also comprising repeating said retransmission requesting step and said if-replacing step until all sub-messages have been correctly received.

156. A method according to any of claims 153 - 155

and also comprising marking each incorrectly received sub-message.

157. A method according to any of claims 153 - 156 and also comprising the step of, prior to transmitting a sub-message, providing error detection code for said sub-message individually, and wherein said error detection code is decoded upon receipt of said sub-message to determine whether or not said sub-message is received correctly.

158. A method according to any of claims 153 - 157 wherein said retransmission requesting step comprises requesting retransmission of the entire message if at least one sub-message is received incorrectly.

159. A method according to any of claims 153 - 158 wherein said step of requesting retransmission comprises sending an acknowledgement when retransmission is no longer necessary, thereby to cause retransmission in the absence of an acknowledgement.

160. A method according to any of claims 153 - 159 and also comprising comparing at least first and second correct transmissions of the same sub-message when said sub-message is received correctly more than once.

161. A method according to claim 160 and also comprising, if the first and second correct transmissions are different, treating said more than once correctly received sub-message as an incorrectly received sub-message until a predetermined stopping criterion is reached.

162. A method according to claim 161 and wherein said predetermined stopping criterion comprises receipt

of a plurality of correct and identical transmissions of said sub-message.

163. A method according to claim 161 and wherein said predetermined stopping criterion comprises receipt of a majority of correct and identical transmissions of said sub-message.

164. A method according to any of claims 153 - 163 and also comprising:

performing a validity check of the message; and

if the validity check fails, repeating the validity check on at least one combination of correct transmissions of the sub-messages contained in said message.

165. A method according to claim 164 and wherein said if-validity check repeating step comprises repeating the validity check on all combinations until one of the combinations yields a successful validity check and requesting retransmission of at least a portion of said message if none of the combinations yields a successful validity check.

166. Apparatus operative to process received messages in order to reduce repeat transmissions of messages, the apparatus comprising:

a message transmitter operative to transmit a message including a plurality of sub-messages;

a retransmission requester operative to request retransmission of at least one incorrectly received sub-message; and

a message replacer, operative, if at least one previously incorrectly received sub-message is correctly received by retransmission, to replace at least one previously incorrectly received sub-message with its

corresponding correctly received sub-message.

167. A method of preventing collisions and coordinating channels between subscriber units in a frequency hopping multiple access communication system wherein a multiplicity of base stations communicate with a multiplicity of subscriber units over a frequency hopping multiple access communication network at a plurality of radio frequencies, the method comprising:

providing, at each subscriber unit, a plurality of frequency channels for transmitting and receiving information signals over slots defined in a time domain and in a frequency domain;

transmitting the information over a plurality of slots; and

allowing each subscriber unit to skip transmission of at least one slot selected in accordance with a predetermined sequence.

168. A method according to claim 167 and comprising:

receiving, at each subscriber unit, the transmitted slots;

recognizing a non-transmitted slot in the slots received; and

building an inactive slot, to replace the non-transmitted slot, by including in the inactive slot a plurality of inactive symbols having imparted a confidence level zero in at least one of a random sequence and an ordered sequence.

169. A method according to either of claim 167 or claim 168 and comprising applying a minimum weight to the inactive symbols during processing of the slots.

170. A method according to any of the claims 167 - 169 and wherein said predetermined sequence is

transmitted to each subscriber unit over a control channel.

171. A communication method wherein a transmitter within a first sector is to transmit to a first subscriber within the first sector, the method comprising:

transmitting to the first subscriber if the first subscriber is located within a fringe area; and

otherwise, determining, for each of at least one time slot, whether or not to transmit from the transmitter to the first subscriber during the at least one time slot.

172. A method according to claim 171 wherein said determining step comprises:

determining, for at least one time slot, whether, during the time slot, there exists a problematic subscriber associated with a neighboring sector who is located within a fringe area between the first and neighboring sectors and who is subject to interference due to transmission from the transmitter to the subscriber; and

if no problematic subscriber exists, transmitting in said time slot.

173. A method according to either of claim 171 or claim 172 wherein said determining step comprises:

defining, for each of a plurality of time segments, a partition of the time during which the transmission occurs, each time segment including at least one time slot, determining whether or not the number of time slots within the time segment in which transmission did not take place exceeds a threshold number of time slots; and

if the threshold is exceeded, transmitting in a

current time slot.

174. A method according to any of the claims 171 – 173 wherein said determining step comprises:

for each of a plurality of positions of a sliding time window including n>1 time slots, determining whether or not the number of time slots within the sliding window, as currently positioned, in which transmission did not take place exceeds a threshold number of time slots; and

if the threshold is exceeded, transmitting in a current time slot.

175. A method according to any of the preceding claims 171 – 174 wherein said determining step comprises determining whether to transmit with a probability p<1 or whether not to transmit.

176. A method according to claim 172 and comprising:

assigning time-slots to each of a plurality of subsectors within the first sector and to each of a plurality of subsectors within the neighboring sector, each sector including central and peripheral subsectors such that the same time-slot is assigned to a peripheral subsector in the first sector and to a central subsector in the neighboring sector; and

assigning more power to downlink transmissions to subscribers within the peripheral subsectors than to downlink transmissions to subscribers within the central subsectors.

177. A method according to claim 172 and also comprising allocating an air resource to the transmitters within the first sector so as to reduce the maximum probability, over the transmitters within the first sector, of existence of a problematic subscriber.

178. A method according to claim 177 wherein said air resource comprises TDMA (time division multiple access) time slots.

179. A method according to claim 177 wherein said air resource comprises FDMA (frequency division multiple access) channels frequencies.

180. A method according to claim 177 wherein said air resource comprises FHMA (frequency hopping multiple access) time/frequency sequences.

181. A method according to any of the preceding claims 176 - 180 and wherein each said time slot comprises an active time slot.

182. A method according to any of the preceding claims 176 - 181 and also comprising, prior to said determining step, the step of transmitting with a probability 1 if the first subscriber is inside said fringe area.

183. A communication method wherein a first subscriber within a first sector is to transmit to a base station, the method comprising:

transmitting to the base station if the first subscriber is not located within a fringe area; and

otherwise, determining, for each of at least one time slots, whether or not to transmit during the time slot.

184. A method according to claim 183 wherein said determining step comprises:

defining, for each of a plurality of time segments, a partition of the time during which the

transmission occurs, each time segment including at least one time slot, determining whether or not the number of time slots within the time segment in which transmission did not take place exceeds a threshold number of the time slots; and

if the threshold is exceeded, transmitting in a current time slot.

185. A method according to either of the claims 183 or 184 wherein said determining step comprises:

for each of a plurality of positions of a sliding time window including n>1 time slots, determining whether or not the number of time slots within the sliding window, as currently positioned, in which transmission did not take place exceeds a threshold number of time slots; and

if the threshold is exceeded, transmitting in a current time slot.

186. A method according to any of the claims 183 - 185 wherein said determining step comprises determining whether to transmit with a probability p<1 or whether not to transmit.

187. A communications system comprising:

apparatus for generating a full-duplex FHMA (frequency hopping multiple access) communication channel between two subscribers; and

apparatus for generating a half-duplex FHMA communication channel between two subscribers who are proximate to one another.

188. A system according to claim 187 and also comprising apparatus for determining, for a given pair of subscribers, whether to generate full-duplex or half-duplex communication and for assigning the given pair of

subscribers to the full-duplex channel generating apparatus or to the half-duplex channel generating apparatus, accordingly.

189. A system according to claim 188 and wherein said apparatus for determining is operative at least partly on a basis of a criterion of the extent of proximity between the given pair of subscribers.

190. A system according to claim 189 and wherein said criterion of the extent of proximity comprises a criterion of signal quality.

191. A system according to any of claims 188 - 190 and wherein said apparatus for determining is operative to assign a given pair of subscribers to the full-duplex channel generating apparatus whenever sufficient air resources are available.

192. A system according to any of claims 188 - 191 and wherein said apparatus for determining is operative to store information associating each subscriber with one of a plurality of talk-groups and wherein at least one pair of subscribers associated with the same talk-group are assigned to the half-duplex channel generating apparatus.

193. A system according to any of claims 187 - 192 and wherein said apparatus for full-duplex channel generating and said apparatus for half-duplex channel generating are located within a base station.

194. A method for generating a half-duplex FHMA communication channel between two subscribers who are proximate to one another within a communications system, the method comprising:

transmitting a channel request from a first subscriber to a second subscriber;

determining a first measure of signal quality of the channel request received at the second subscriber;

sending a request acknowledgement from the second subscriber to the first subscriber;

determining a second measure of signal quality of the request acknowledgement received by the first subscriber; and

generating a half-duplex FHMA communication channel between the first subscriber and the second subscriber only if both the first measure of signal quality and the second measure of signal quality meet a predetermined criterion.

195. A method according to claim 194 and wherein the generating step comprises:

reversing use of an uplink channel and a downlink channel in exactly one subscriber from among the first and second subscribers.

196. A method according to claim 194 and wherein the generating step comprises:

using exactly one of an uplink channel and a downlink channel for both transmission and reception in both of the first and second subscribers.

197. A communications system comprising:

apparatus for generating a mediated FHMA (frequency hopping multiple access) communication channel between two subscribers; and

apparatus for generating a direct FHMA communication channel between two subscribers who are proximate to one another.

198. A system according to claim 197 and also

comprising apparatus for determining, for a given pair of subscribers, whether to generate mediated or direct communication and for assigning the given pair of subscribers to the mediated channel generating apparatus or to the direct channel generating apparatus, accordingly.

199. A system according to claim 198 and wherein said apparatus for determining is operative at least partly on a basis of a criterion of the extent of proximity between the given pair of subscribers.

200. A system according to claim 199 and wherein said criterion of the extent of proximity comprises a criterion of signal quality.

201. A system according to any of claims 198 - 200 and wherein said apparatus for determining is operative to assign a given pair of subscribers to the mediated channel generating apparatus whenever sufficient air resources are available.

202. A system according to any of claims 198 - 201 and wherein said apparatus for determining is operative to store information associating each subscriber with one of a plurality of talk-groups and wherein at least one pair of subscribers associated with the same talk-group are assigned to the direct channel generating apparatus.

203. A system according to any of claims 197 - 202 and wherein said apparatus for mediated channel generating and said apparatus for direct channel generating are located within a base station.

204. A system according to any of claims 197 - 203 and wherein the direct FHMA communication channel

comprises a full-duplex communication channel.

205. A system according to any of claims 197 - 203 and wherein the direct FHMA communication channel comprises a half-duplex communication channel.

206. A method for generating a direct FHMA communication channel between two subscribers who are proximate to one another within a communications system, the method comprising:

transmitting a channel request from a first subscriber to a second subscriber;

determining a first measure of signal quality of the channel request received by the second subscriber;

sending a request acknowledgement from the second subscriber to the first subscriber;

determining a second measure of signal quality of the request acknowledgement received at the first subscriber; and

generating a direct FHMA communication channel between the first subscriber and the second subscriber only if both the first measure of signal quality and the second measure of signal quality meet a predetermined criterion.

207. A method according to claim 206 and wherein the generating step comprises:

choosing one of the first subscriber unit and the second subscriber unit;

reversing use of an uplink channel and a downlink channel in the chosen subscriber unit.

208. A method according to claim 206 and wherein the generating step comprises:

using exactly one of an uplink channel and a downlink channel for both transmission and reception in

both of the first and second subscribers.

209. A method according to any of claims 206 - 208 and wherein the direct FHMA communication channel comprises a full-duplex communication channel.

210. A method according to any of claims 206 - 208 and wherein the direct FHMA communication channel comprises a half-duplex communication channel.

211. A communications method comprising:

generating a full-duplex FHMA (frequency hopping multiple access) communication channel between two subscribers; and

generating a half-duplex FHMA communication channel between two subscribers who are proximate to one another.

212. Apparatus for generating a half-duplex FHMA communication channel between two subscribers who are proximate to one another within a communications system, the apparatus comprising:

a channel requester operative to transmit a channel request from a first subscriber to a second subscriber;

a first signal quality determining apparatus operative to determine a first measure of signal quality of the channel request received at the second subscriber;

a request acknowledger operative to send a request acknowledgement from the second subscriber to the first subscriber;

second signal quality determining apparatus operative to determine a second measure of signal quality of the request acknowledgement received by the first subscriber; and

a half-duplex FHMA generator operative to

generate a half-duplex FHMA communication channel between the first subscriber and the second subscriber only if both the first measure of signal quality and the second measure of signal quality meet a predetermined criterion.

213. A communications method comprising:

generating a mediated FHMA (frequency hopping multiple access) communication channel between two subscribers; and

generating a direct FHMA communication channel between two subscribers who are proximate to one another.

214. Apparatus for generating a direct FHMA communication channel between two subscribers who are proximate to one another within a communications system, the apparatus comprising:

a channel requester operative to transmit a channel request from a first subscriber to a second subscriber;

a first signal quality determining apparatus operative to determine a first measure of signal quality of the channel request received by the second subscriber;

a request acknowledger operative to send a request acknowledgement from the second subscriber to the first subscriber;

a second signal quality determining apparatus operative to determine a second measure of signal quality of the request acknowledgement received at the first subscriber; and

a FHMA communication channel generator operative to generate a direct FHMA communication channel between the first subscriber and the second subscriber only if both the first measure of signal quality and the second measure of signal quality meet a predetermined criterion.

215. A system according to claim 187 where the apparatus for generating a half-duplex FHMA communication channel between two subscribers who are proximate to one another comprises apparatus for generating a half-duplex FHMA communication channel between two subscribers who are geographically proximate to one another.

216. A method for reducing interference in a communication system between a plurality of sectors in a vicinity of a base station including a fringe area of at least two of the sectors, the method comprising:

assigning a surrounding area which surrounds the base station to an individual one of said plurality of communication system sectors; and

allocating air resources to communication system subscriber units within said surrounding area, thereby to allow communication between at least one of said subscriber units and at least one second party.

217. A method according to claim 216 and wherein said communication system comprises an FHMA (frequency hopping multiple access) communication system.

218. Base station antenna apparatus comprising:

a plurality of sector antennas disposed at a height H relative to the ground; and

an auxiliary antenna disposed at a height $h < H$ relative to the ground having a radiation pattern which includes an entire azimuthal vicinity of the base station.

219. Base station apparatus according to claim 218 and wherein said sector antennas are located directly above the auxiliary antenna.

220. Apparatus according to claim 218 and wherein

said sector antennas are not located directly above the auxiliary antenna.

221. Apparatus according to any of claims 218 - 220 and wherein said auxiliary antenna is fed by a dedicated transmitter.

222. Apparatus according to any of claims 218 - 221 wherein said auxiliary antenna comprises an omnidirectional antenna.

223. Apparatus operative to reduce interference in a communication system between a plurality of sectors in a vicinity of a base station including a fringe area of at least two of the sectors, the apparatus comprising:

a sector assigner operative to assign a surrounding area which surrounds the base station to an individual one of said plurality of sectors; and

an air resource allocator operative to allocate air resources to subscriber units within said surrounding area, thereby allowing communication between at least one of said subscriber units and at least one second party.

224. A base station antenna construction method comprising:

disposing a plurality of sector antennas at a height H relative to the ground; and

disposing at a height h < H relative to the ground an auxiliary antenna having a radiation pattern which includes an entire azimuthal vicinity of the base station.

225. A method of deriving the state of a channel on which communication signals are received, comprising the steps of:

receiving and demodulating a communication

signal;

determining the modulation point of the communication signal as transmitted;

determining the in-phase component of the received communication signal;

determining the quadrature component of the received communication signal; and

determining the channel state of the channel on which the communication signal was received from the ratio of the in-phase component to the quadrature component.

226. Apparatus for deriving the state of a channel on which communication signals are received, comprising:

means for receiving and demodulating a communication signal;

means for determining the modulation point of the communication signal as transmitted;

means for determining the in-phase component of the received communication signal;

means for determining the quadrature component of the received communication signal; and

means for determining the channel state of the channel on which the communication signal was received from the ratio of the in-phase component to the quadrature component.

227. A method for deriving the state of a channel on which communication signals are received, comprising the steps of:

receiving and demodulating a plurality of communication signals;

determining the modulation point of each communication signal;

determining a in-phase component of each communication signal;

summing the value of the in-phase components;

determining a quadrature component of each communication signal;

summing the value of the quadrature components; and

determining the channel state of the channel on which the plurality of communication signals was received from the ratio of the summed value of the in-phase components to the summed value of the absolute value of the quadrature components.

228. The method of claim 227, wherein the plurality of communication signals used are from a time slot.

229. The method of claim 228, wherein the plurality of communication signals within a time slot are transmitted over a single frequency channel.

230. The method of claim 229, wherein the channel state is determined in accordance with the following equation:

$$CS = \frac{\sum_{Slot} \mathrm{Real}\,(Si)}{\sum_{Slot} \left| \mathrm{Imag}\,(Si) \right|}$$

231. Apparatus for deriving the state of a channel on which communication signals are received, comprising:

means for receiving and demodulating a plurality of communication signals;

means for determining the modulation point of each communication signal;

means for determining an in-phase component of

each communication signal;

means for summing the value of the in-phase components;

means for determining a quadrature component of each communication signal;

means for summing the value of the quadrature components; and

means for determining the channel state of the channel on which the plurality of communication signals was received from the ratio of the summed value of the in-phase components to the summed value of the absolute value of the quadrature components.

232. The apparatus of claim 231, wherein the plurality of communication signals used are from a time slot.

233. The apparatus of claim 232, wherein the plurality of communication signals within a time slot are transmitted over a single frequency channel.

234. The apparatus of claim 233, wherein the channel state is determined in accordance with the following equation:

$$CS = \frac{\sum_{Slot} \text{Real}(Si)}{\sum_{Slot} |\text{Imag}(Si)|}$$

235. A method for deriving the state of a channel on which communication signals are received, comprising:

receiving a plurality of communication signals;

determining the average of the in-phase

components of the received communication signals;

determining the average of the quadrature components of the received communication signals; and

determining the channel state of the channel on which the plurality of communication signals was received from the ratio of the average value of the in-phase components to the average of the quadrature components.

236. The method of claim 235, wherein the plurality of communication signals that are received come from a time slot.

237. Apparatus for deriving the state of a channel on which communication signals are received, comprising:

means for receiving a plurality of communication signals;

means for determining the average of the in-phase components of the received communication signals;

means for determining the average of the quadrature components of the received communication signals; and

means for determining the channel state of the channel on which the plurality of communication signals was received from the ratio of the average value of the in-phase components to the average of the quadrature components.

238. The apparatus of claim 237, wherein the plurality of communication signals that are received come from a time slot.

239. A method of processing communication signals, comprising the steps of:

determining the channel state when a communication signal is received; and

erasing the signal if the channel state is not

better than a predetermined level.

240. The method of claim 239, wherein the channel state is determined using a plurality of communication signals and the plurality of communication signals is erased if the channel state is not better than the predetermined level.

241. A method of processing communication signals in a time slot, comprising the steps of:

determining the channel state of the communication signals received in the time slot; and

erasing the communication signals in the time slot if the channel state is not better than a predetermined level.

242. Apparatus for processing communication signals, comprising:

means for determining the channel state when a communication signal is received; and

means for erasing the signal if the channel state is not better than a predetermined level.

243. The apparatus of claim 242, wherein the channel state is determined using a plurality of communication signals and the plurality of communication signals is erased if the channel state is not better than the predetermined level.

244. Apparatus for processing communication signals in a time slot, comprising:

means for determining the channel state of the communication signals received in the time slot; and

means for erasing the communication signals in the time slot if the channel state is not better than a predetermined level.

245. A method for receiving and processing a first communication signal and a second communication signal received at the same time, comprising the steps of:

determining a first channel state when the first communication signal is received;

determining a second channel state when the second communication signal is received; and

selecting one of the two communication signals based on the first and the second channel states.

246. The method of claim 245, wherein the communication signal associated with the better channel state is selected.

247. A method for processing a first set of communication signals received in a first time slot on a first antenna and a second set of communication signals received on a second antenna at the same time, comprising the steps of:

determining a first channel state associated with the time slot when the first set of communication signals is received;

determining a second channel state associated with the time slot when the second set of communication signals is received; and

selecting one of the two communication signals based on the first and the second channel states.

248. The method of claim 247, wherein the set of communication signals associated with the better channel state is selected.

249. Apparatus for receiving and processing a first communication signal and a second communication signal received at the same time, comprising:

means for determining a first channel state when the first communication signal is received;

means for determining a second channel state when the second communication signal is received; and

means for selecting one of the two communication signals based on the first and the second channel states.

250. The apparatus of claim 249, wherein the communication signal associated with the better channel state is selected.

251. Apparatus for processing a first set of communication signals received in a first time slot on a first antenna and a second set of communication signals received on a second antenna at the same time, comprising:

means for determining a first channel state associated with the time slot when the first set of communication signals is received;

means for determining a second channel state associated with the time slot when the second set of communication signals is received; and

means for selecting one of the two communication signals based on the first and the second channel states.

252. The apparatus of claim 251, wherein the set of communication signals associated with the better channel state is selected.

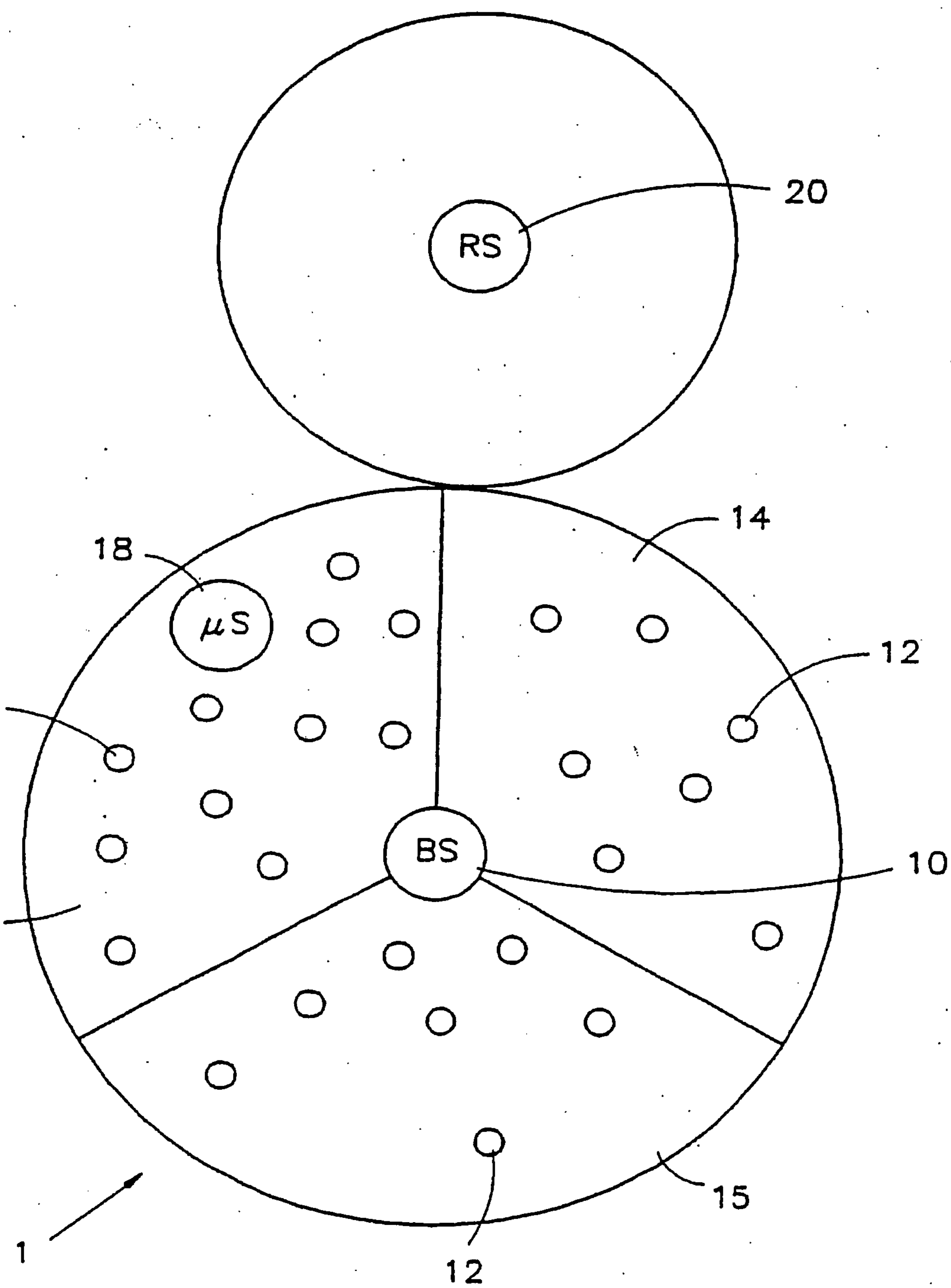


FIG. 1

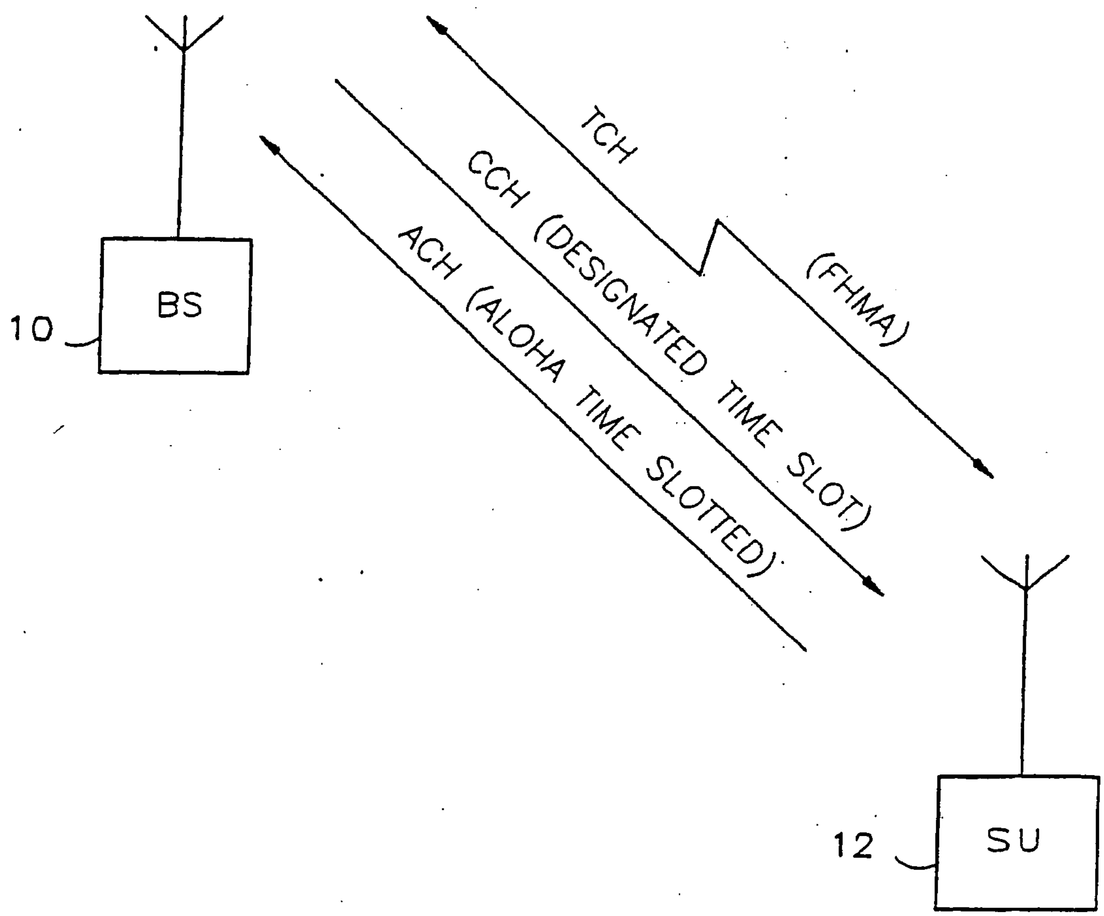

BS
10
TCH
(FHMA)
CCH (DESIGNATED TIME SLOT)
ACH (ALOHA TIME SLOTTED)
SU
12

FIG. 2

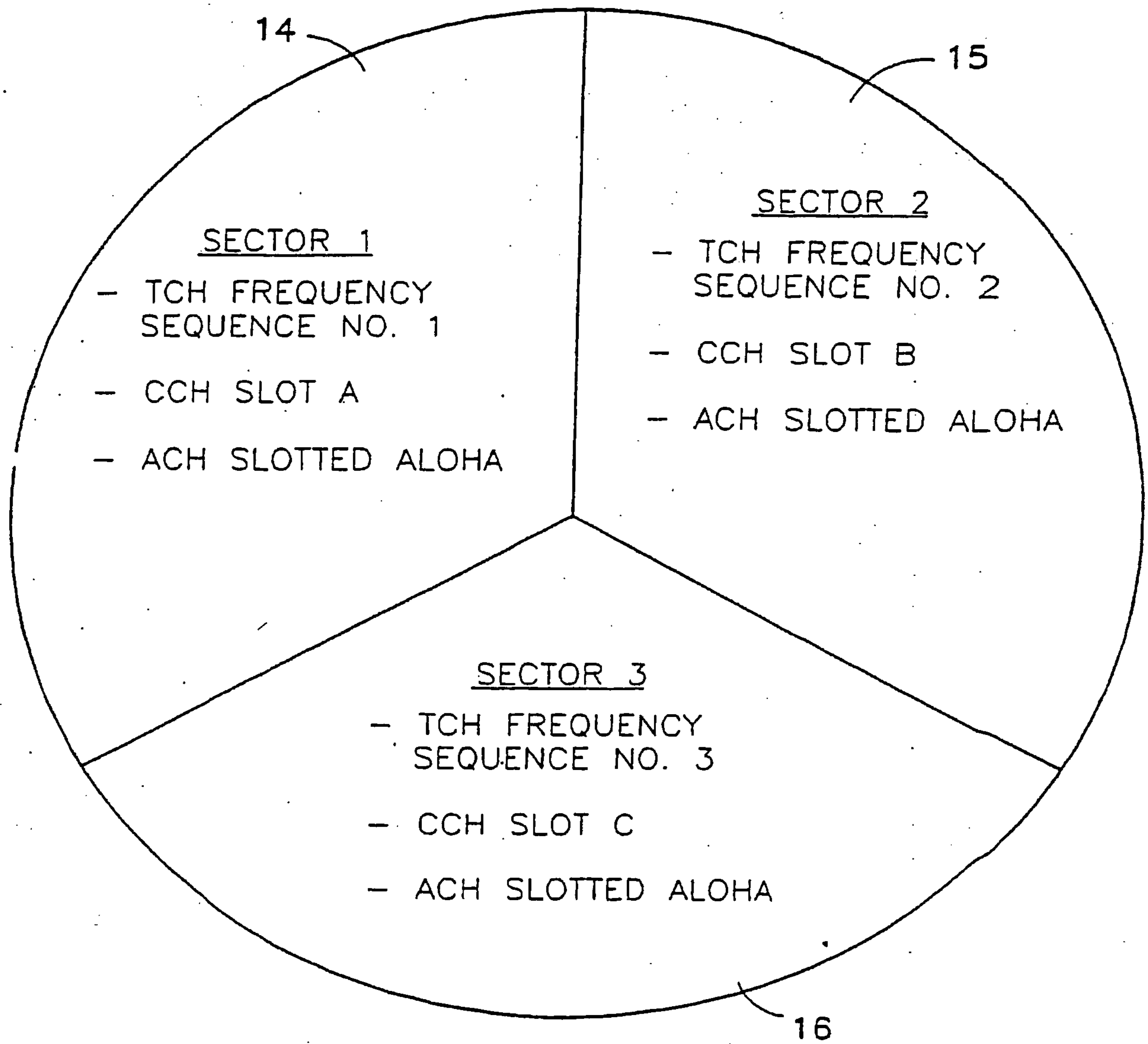

14
15
SECTOR 1
– TCH FREQUENCY SEQUENCE NO. 1
– CCH SLOT A
– ACH SLOTTED ALOHA
SECTOR 2
– TCH FREQUENCY SEQUENCE NO. 2
– CCH SLOT B
– ACH SLOTTED ALOHA
SECTOR 3
– TCH FREQUENCY SEQUENCE NO. 3
– CCH SLOT C
– ACH SLOTTED ALOHA
16

FIG. 3A

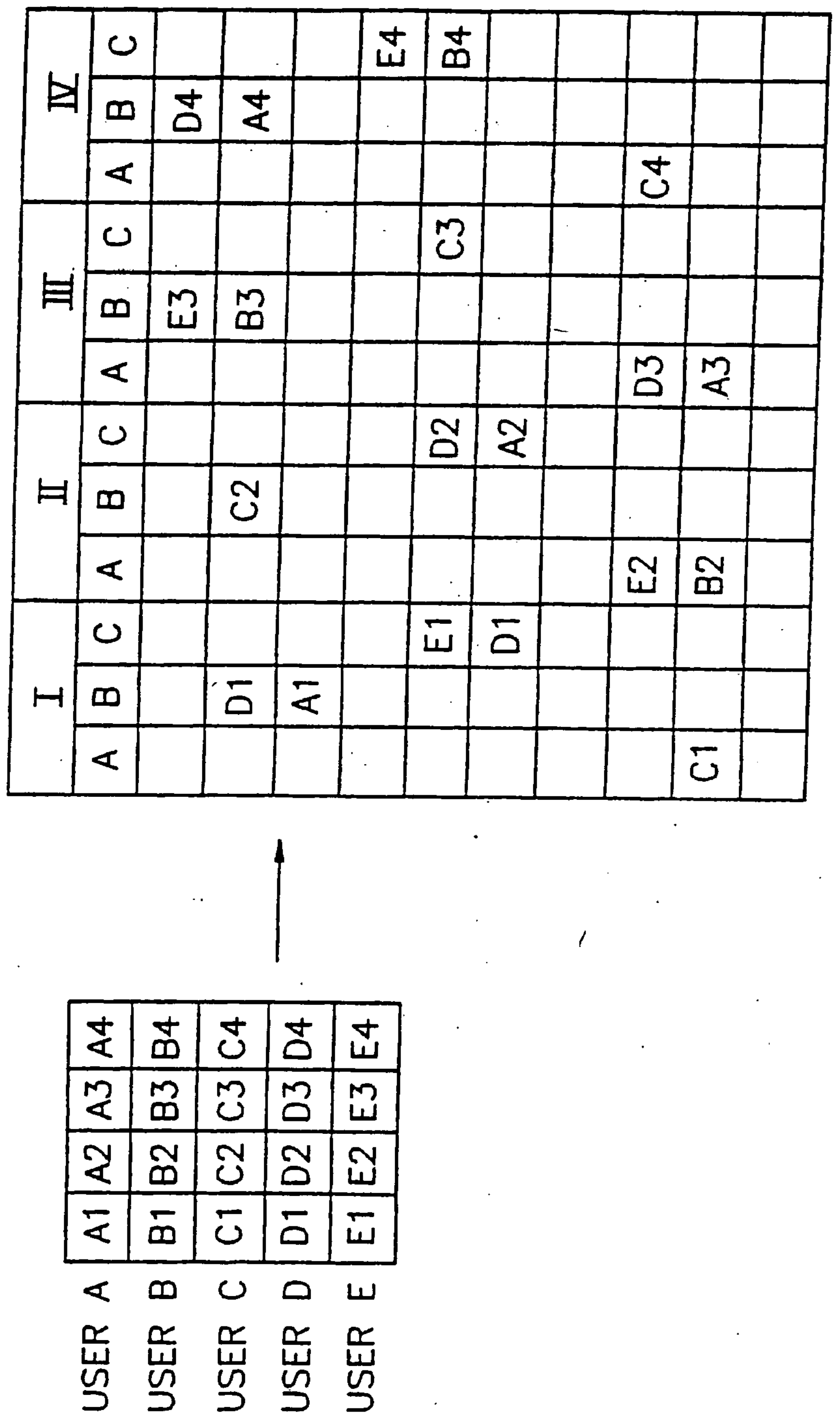
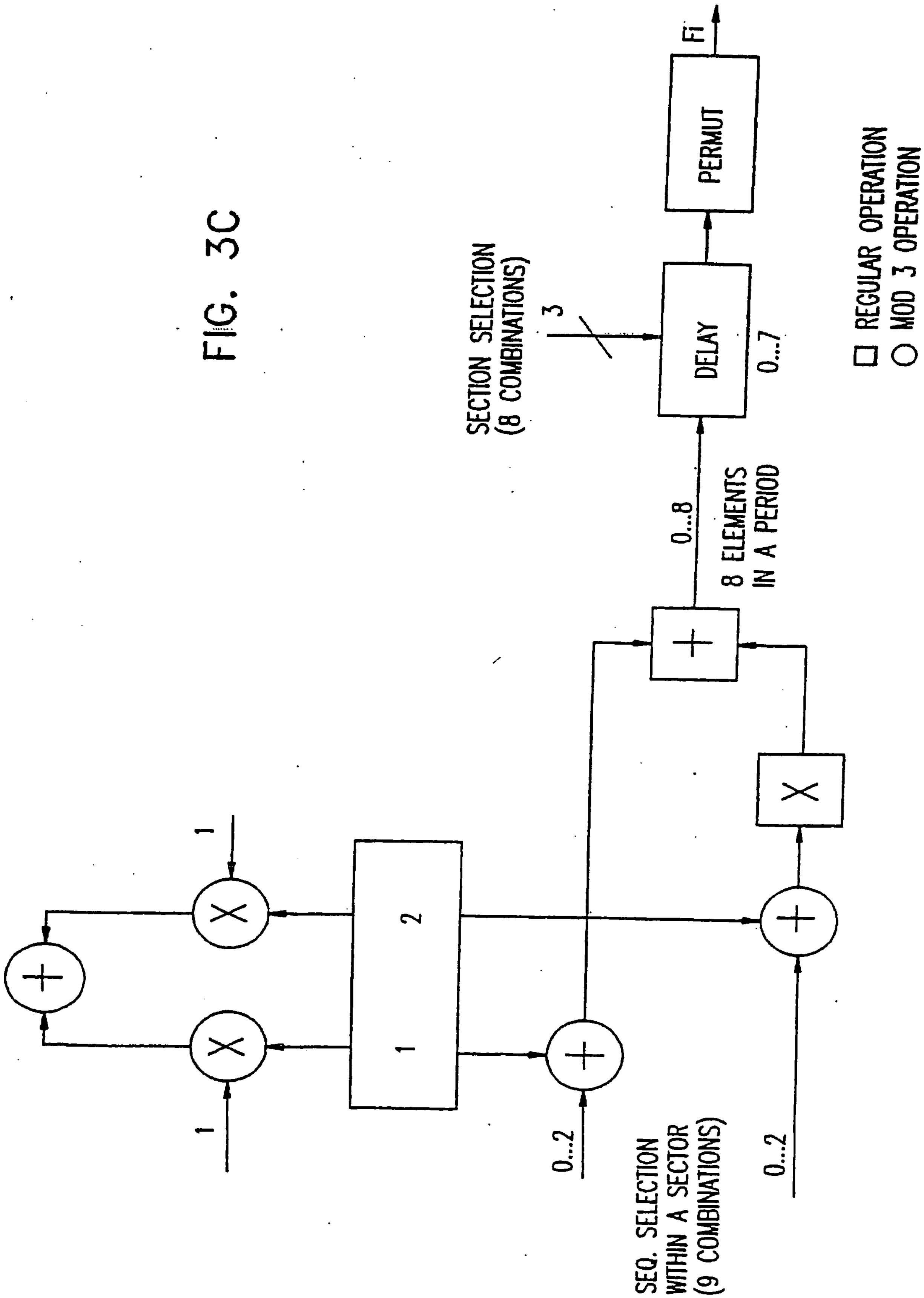


FIG. 3B

FIG. 3C


SECTION SELECTION
(8 COMBINATIONS)
3
DELAY
0...7
PERMUT
Fi
□ REGULAR OPERATION
○ MOD 3 OPERATION
0...8
8 ELEMENTS
IN A PERIOD
1
2
1
1
0...2
0...2
SEQ. SELECTION
WITHIN A SECTOR
(9 COMBINATIONS)

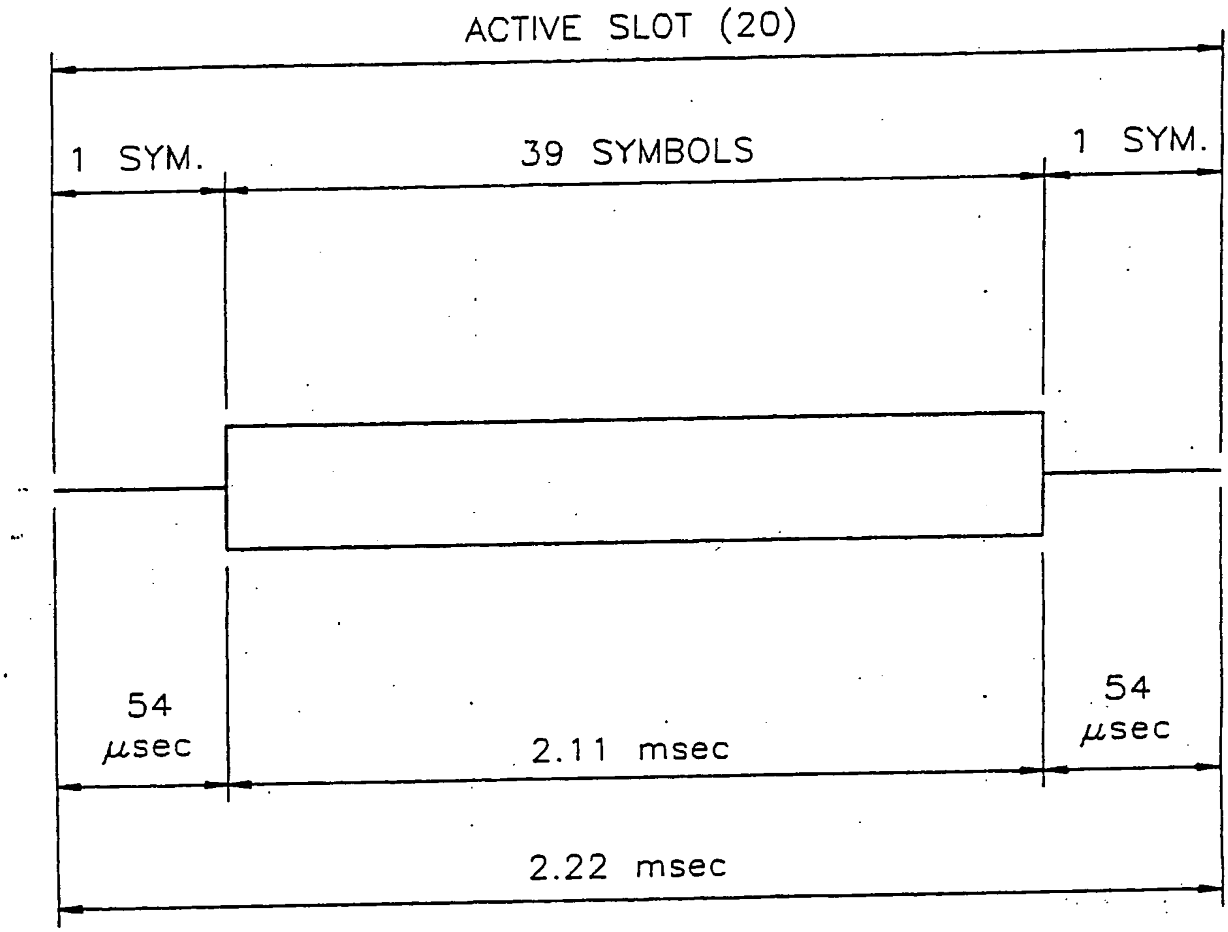

ACTIVE SLOT (20)
1 SYM.
39 SYMBOLS
1 SYM.
54 μsec
2.11 msec
54 μsec
2.22 msec

FIG. 4

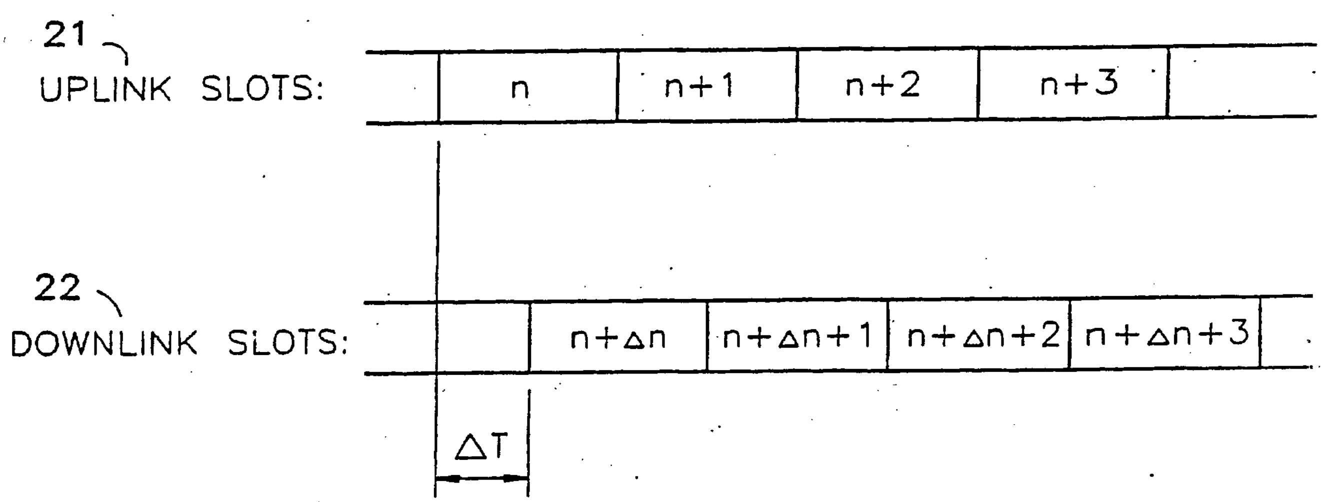

21
UPLINK SLOTS:
n
n+1
n+2
n+3
22
DOWNLINK SLOTS:
n+Δn
n+Δn+1
n+Δn+2
n+Δn+3
ΔT

FIG. 5

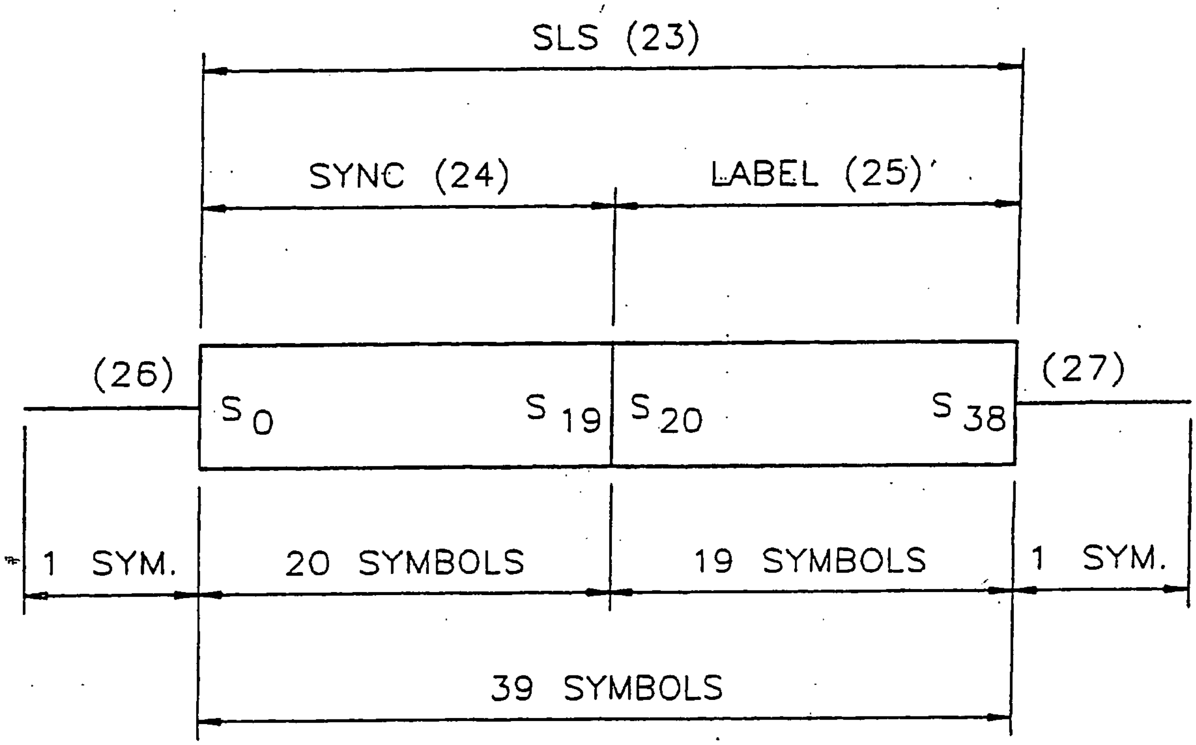

SLS (23)
SYNC (24)
LABEL (25)
(26)
(27)
S 0
S 19
S 20
S 38
1 SYM.
20 SYMBOLS
19 SYMBOLS
1 SYM.
39 SYMBOLS

FIG. 6

FIG. 7

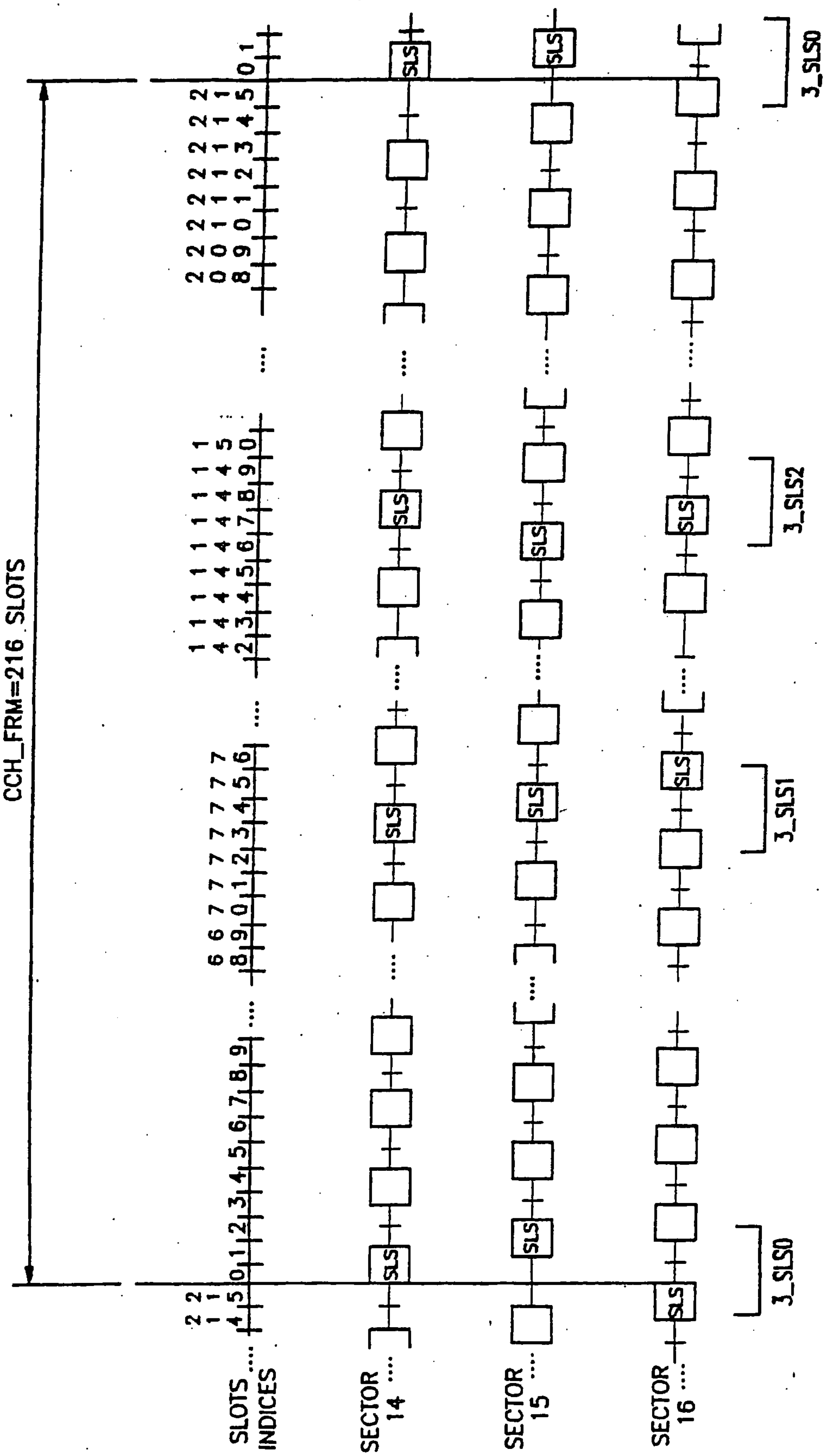
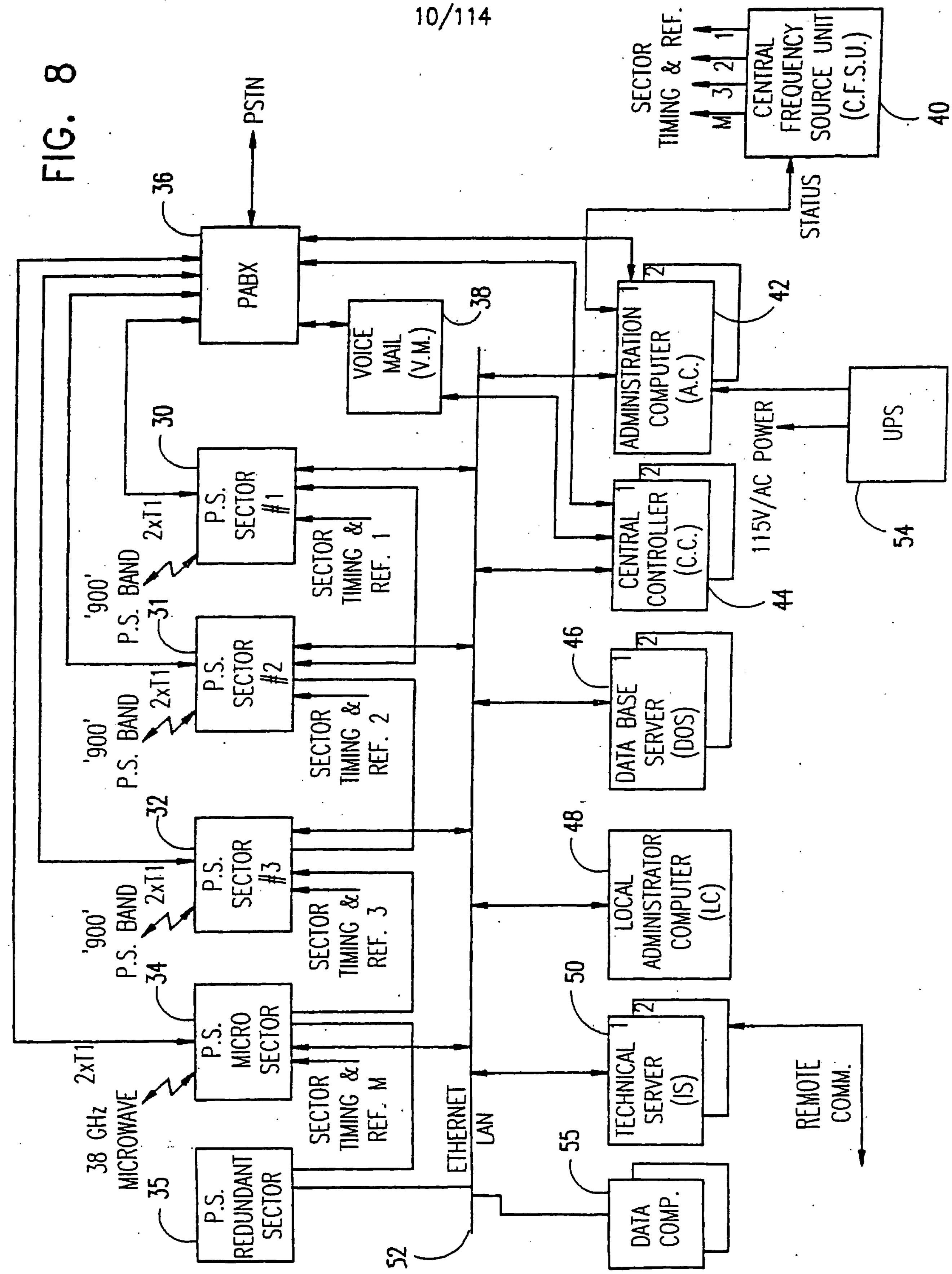

CCH_FRM=216 SLOTS
SLOTS INDICES
SECTOR 14
SECTOR 15
SECTOR 16
SLS
3_SLS0
3_SLS1
3_SLS2

FIG. 8


PSTN
36
PABX
VOICE MAIL (V.M.)
38
SECTOR TIMING & REF.
M 3 2 1
CENTRAL FREQUENCY SOURCE UNIT (C.F.S.U.)
40
STATUS
ADMINISTRATION COMPUTER (A.C.)
1
2
42
UPS
54
115V/AC POWER
CENTRAL CONTROLLER (C.C.)
44
30
P.S. SECTOR #1
2xT1
'900' P.S. BAND
SECTOR TIMING & REF. 1
31
P.S. SECTOR #2
SECTOR TIMING & REF. 2
46
DATA BASE SERVER (DOS)
32
P.S. SECTOR #3
SECTOR TIMING & REF. 3
48
LOCAL ADMINISTRATOR COMPUTER (LC)
34
P.S. MICRO SECTOR
38 GHz MICROWAVE
SECTOR TIMING & REF. M
50
TECHNICAL SERVER (IS)
REMOTE COMM.
ETHERNET LAN
35
P.S. REDUNDANT SECTOR
55
DATA COMP.
52

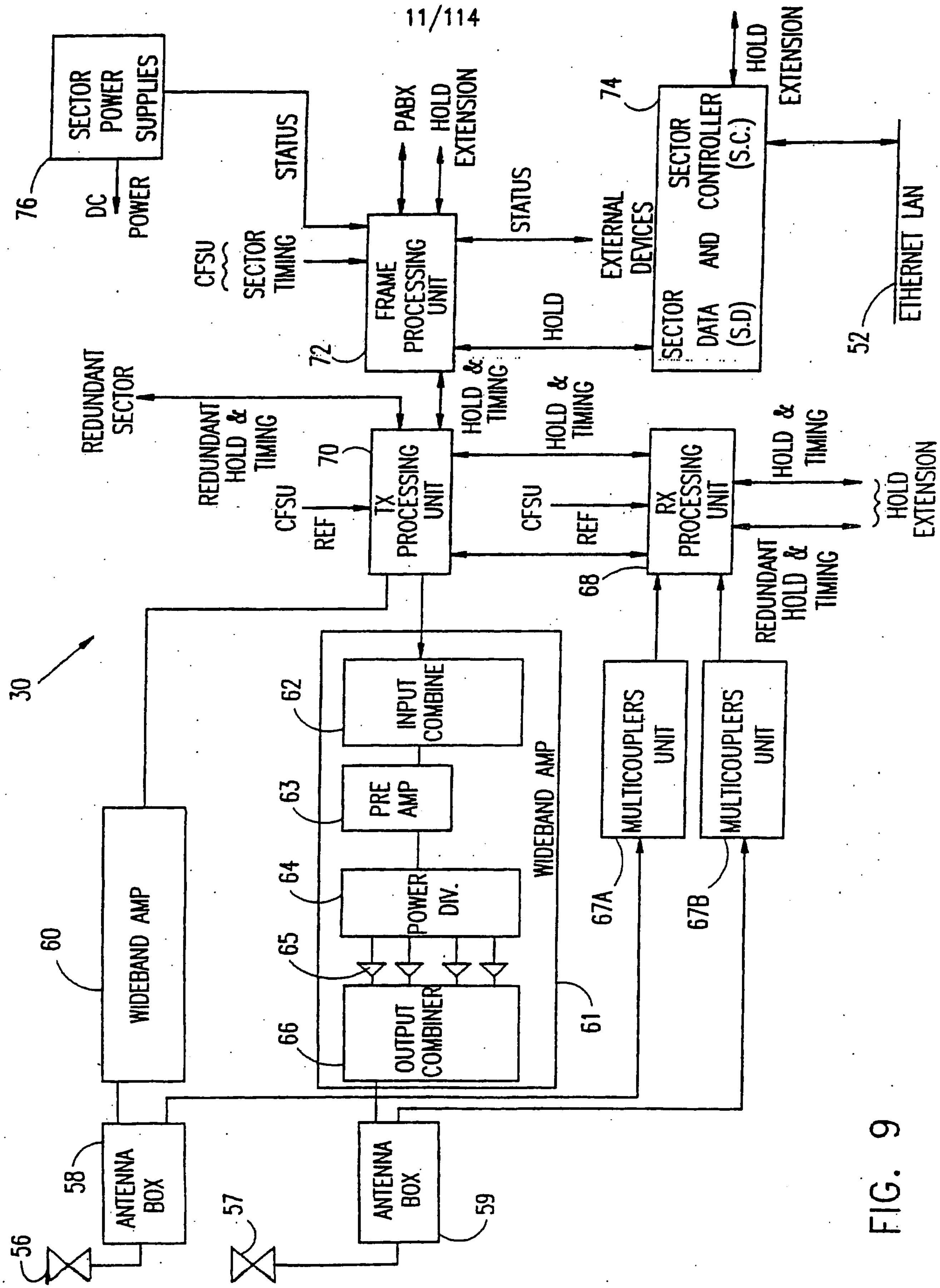

SECTOR POWER SUPPLIES
76
DC POWER
STATUS
PABX
HOLD EXTENSION
74
HOLD EXTENSION
SECTOR CONTROLLER (S.C.)
SECTOR DATA AND (S.D)
ETHERNET LAN
52
CFSU
SECTOR TIMING
FRAME PROCESSING UNIT
72
STATUS
EXTERNAL DEVICES
HOLD
HOLD & TIMING
REDUNDANT SECTOR
REDUNDANT HOLD & TIMING
70
CFSU
REF
TX PROCESSING UNIT
HOLD & TIMING
CFSU
REF
RX PROCESSING UNIT
68
HOLD & TIMING
HOLD EXTENSION
REDUNDANT HOLD & TIMING
30
62
INPUT COMBINE
63
PRE AMP
64
POWER DIV.
65
66
OUTPUT COMBINER
WIDEBAND AMP
61
MULTICOUPLERS UNIT
67A
MULTICOUPLERS UNIT
67B
60
WIDEBAND AMP
58
ANTENNA BOX
56
57
ANTENNA BOX
59

FIG. 9

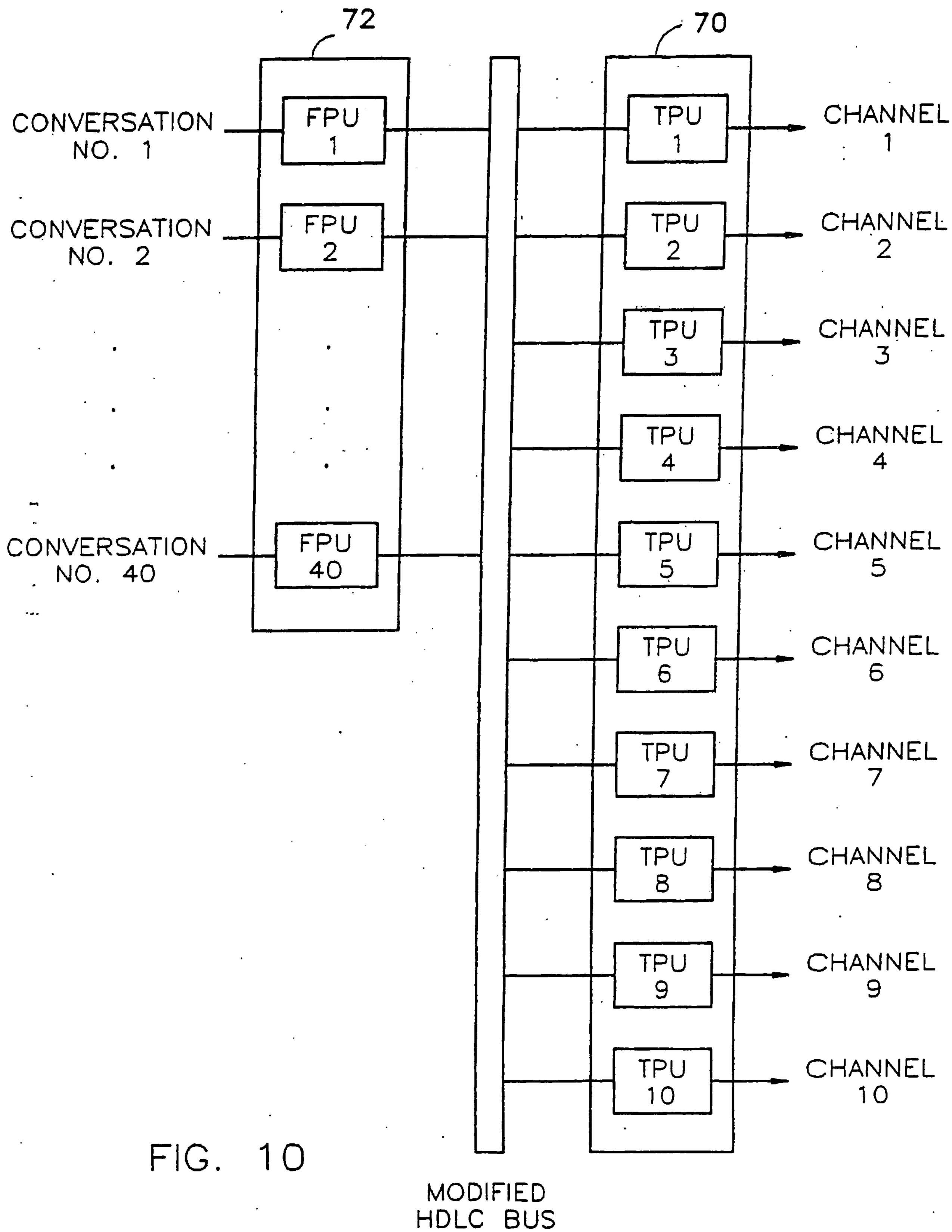

72
70
CONVERSATION NO. 1
CONVERSATION NO. 2
CONVERSATION NO. 40
FPU 1
FPU 2
FPU 40
TPU 1
TPU 2
TPU 3
TPU 4
TPU 5
TPU 6
TPU 7
TPU 8
TPU 9
TPU 10
CHANNEL 1
CHANNEL 2
CHANNEL 3
CHANNEL 4
CHANNEL 5
CHANNEL 6
CHANNEL 7
CHANNEL 8
CHANNEL 9
CHANNEL 10
MODIFIED HDLC BUS

FIG. 10

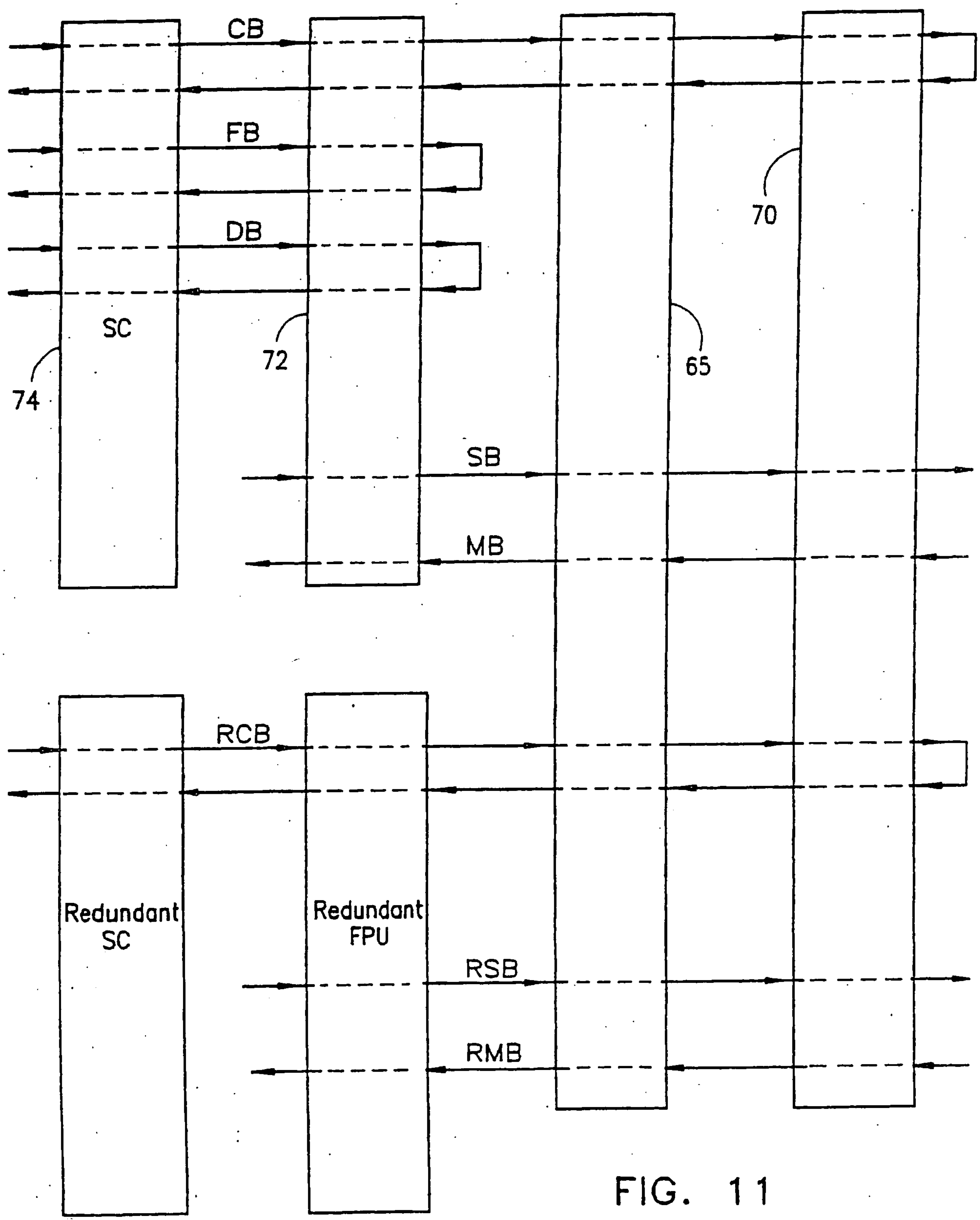

CB
FB
DB
SC
74
72
65
70
SB
MB
RCB
Redundant SC
Redundant FPU
RSB
RMB

FIG. 11

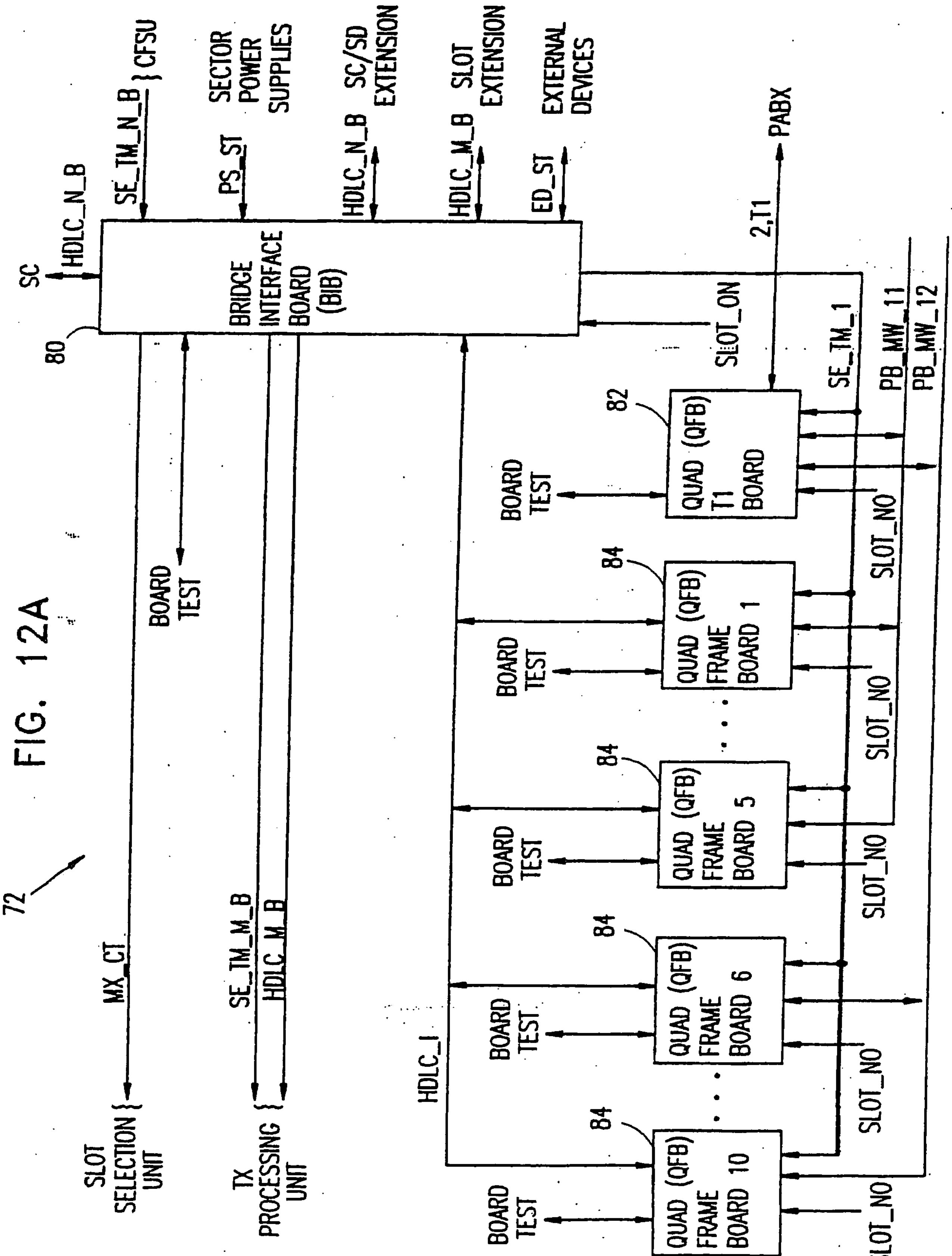
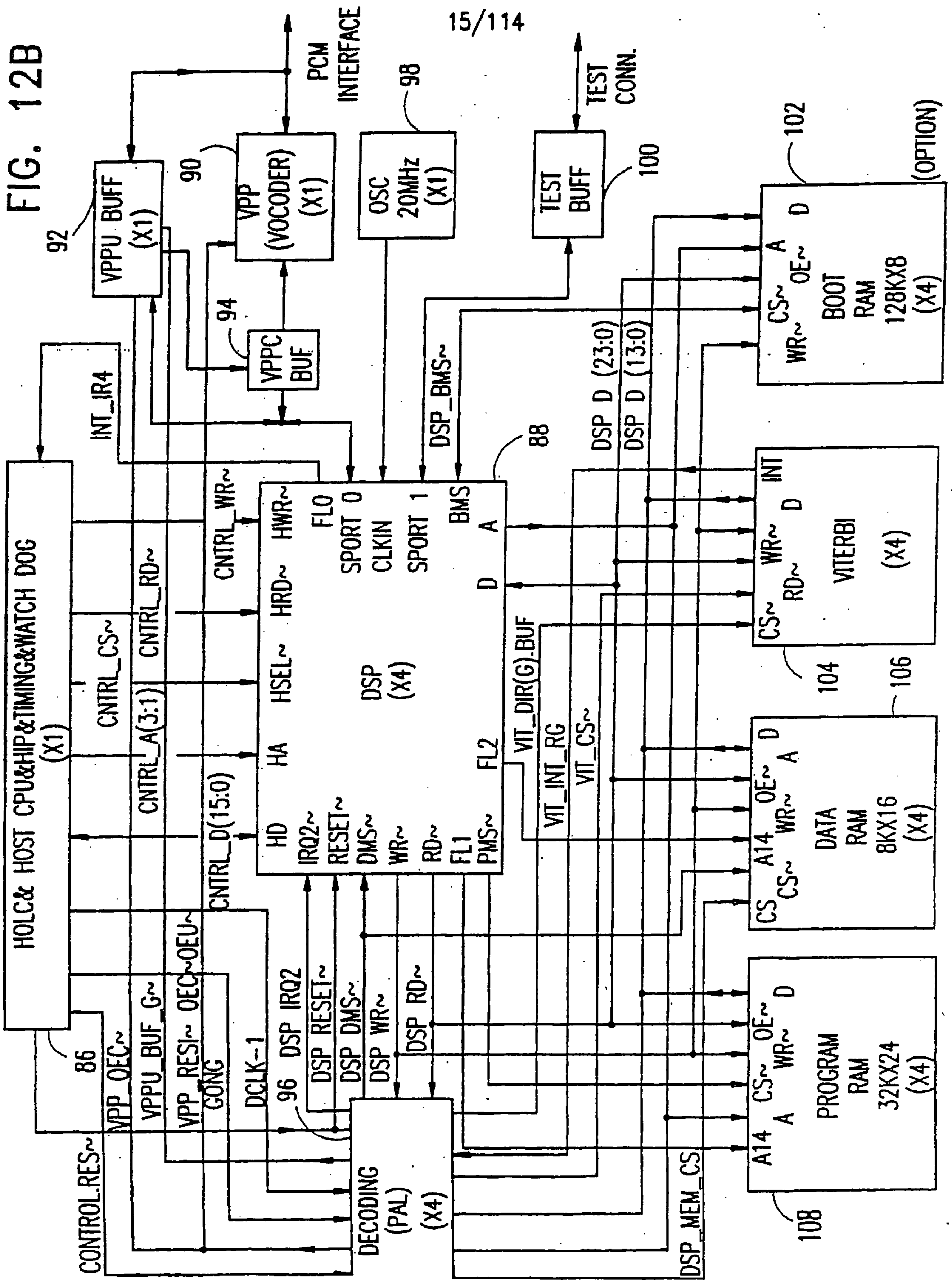

72
80
SC
HDLC_N_B
SE_TM_N_B
CFSU
PS_ST
SECTOR POWER SUPPLIES
HDLC_N_B
SC/SD EXTENSION
HDLC_M_B
SLOT EXTENSION
ED_ST
EXTERNAL DEVICES
BRIDGE INTERFACE BOARD (BIB)
2,T1
PABX
SLOT_ON
SE_TM_1
PB_MW_11
PB_MW_12
BOARD TEST
82
QUAD (QFB) T1 BOARD
84
QUAD (QFB) FRAME BOARD 1
QUAD (QFB) FRAME BOARD 5
QUAD (QFB) FRAME BOARD 6
QUAD (QFB) FRAME BOARD 10
SLOT_NO
MX_CT
SE_TM_M_B
HDLC_M_B
HDLC_I
SLOT SELECTION UNIT
TX PROCESSING UNIT

FIG. 12A

FIG. 12B


PCM INTERFACE
98
TEST CONN.
92
VPPU BUFF (X1)
90
VPP (VOCODER) (X1)
OSC 20MHz (X1)
TEST BUFF
100
102
(OPTION)
BOOT RAM 128KX8 (X4)
CS~ WR~ OE~ A D
94
VPPC BUF
INT_IR4
DSP_BMS~
DSP D (23:0)
DSP D (13:0)
88
HOLC& HOST CPU&HIP&TIMING&WATCH DOG (X1)
CNTRL_WR~
CNTRL_RD~
CNTRL_CS~
CNTRL_A(3:1)
CNTRL_D(15:0)
HWR~ HRD~ HSEL~ HA HD
FL0 SPORT 0 CLKIN SPORT 1 BMS A D
DSP (X4)
IRQ2~ RESET~ DMS~ WR~ RD~ FL1 PMS~ FL2
VIT_DIR(G).BUF
VIT_INT_RG
VIT_CS~
INT D WR~ RD~ CS~
VITERBI (X4)
104
106
D A OE~ WR~ A14 CS~ CS
DATA RAM 8KX16 (X4)
D OE~ WR~ CS~ A A14
PROGRAM RAM 32KX24 (X4)
108
86
VPP_OEC~
VPPU_BUF_G~
VPP_RESI~ OEC~OEU~
GONG
DCLK-1
96
DSP IRQ2
DSP RESET~
DSP DMS~
DSP WR~
DSP RD~
CONTROL.RES~
DECODING (PAL) (X4)
DSP_MEM_CS

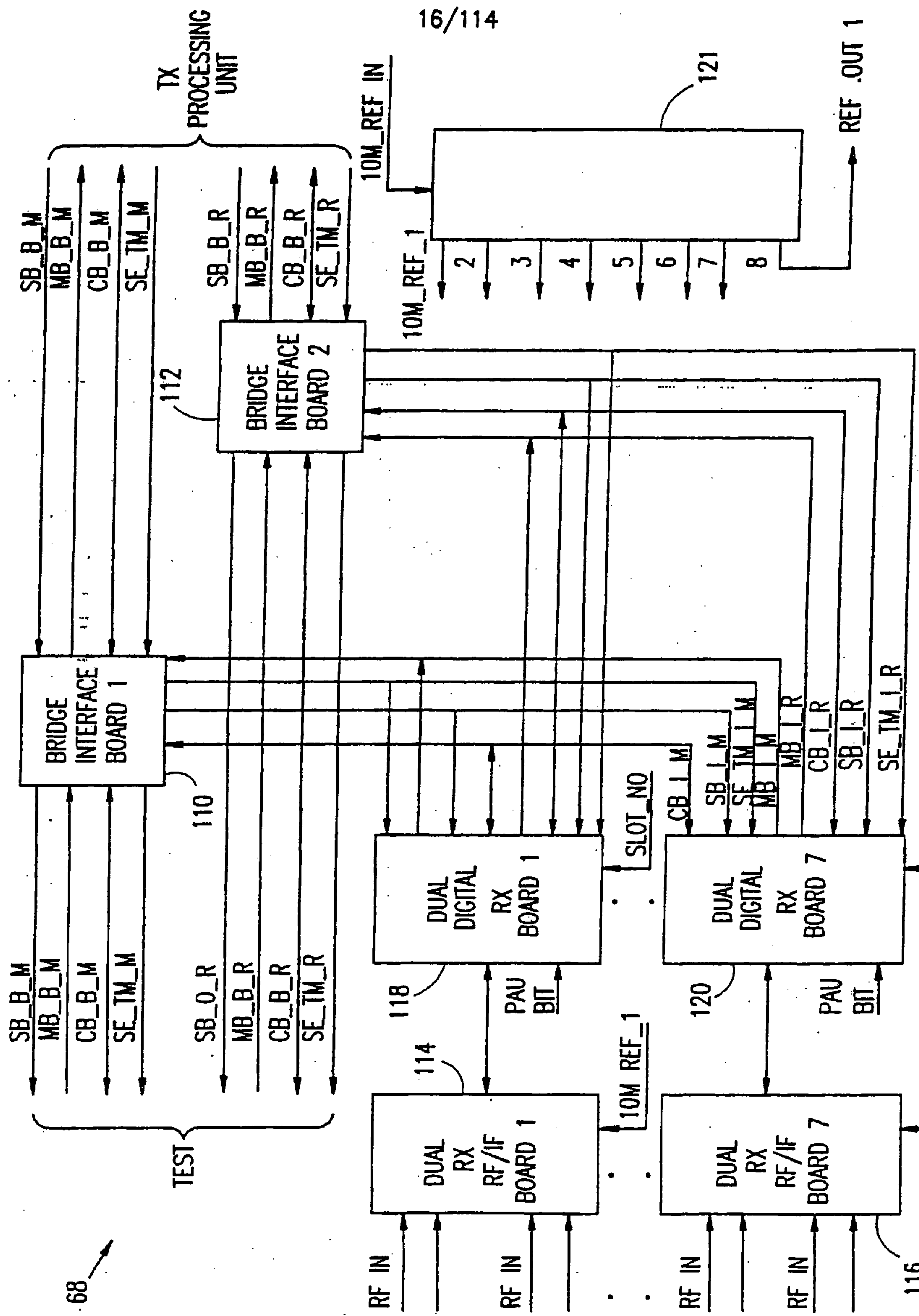

TX PROCESSING UNIT
SB_B_M
MB_B_M
CB_B_M
SE_TM_M
SB_B_R
MB_B_R
CB_B_R
SE_TM_R
10M_REF IN
121
REF .OUT 1
10M_REF_1
2
3
4
5
6
7
8
112
BRIDGE INTERFACE BOARD 2
BRIDGE INTERFACE BOARD 1
110
CB_I_M
SB_I_M
SE_TM_I_M
MB_I_M
MB_I_R
CB_I_R
SB_I_R
SE_TM_I_R
SLOT_NO
DUAL DIGITAL RX BOARD 1
DUAL DIGITAL RX BOARD 7
118
120
PAU
BIT
SB_B_M
MB_B_M
CB_B_M
SE_TM_M
SB_O_R
MB_B_R
CB_B_R
SE_TM_R
TEST
114
10M REF_1
10M REF_7
DUAL RX RF/IF BOARD 1
DUAL RX RF/IF BOARD 7
RF IN
116
68

FIG. 13A

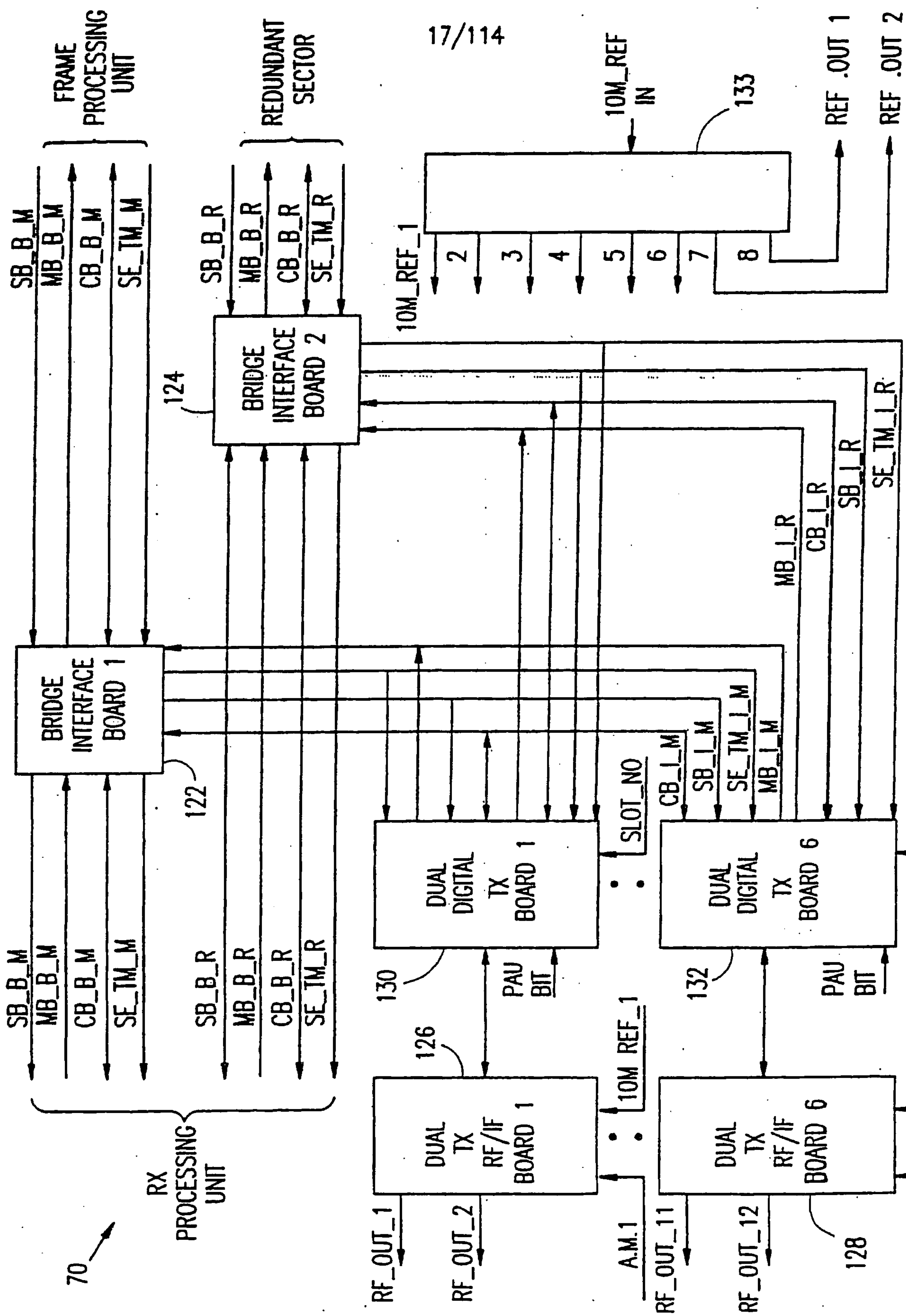
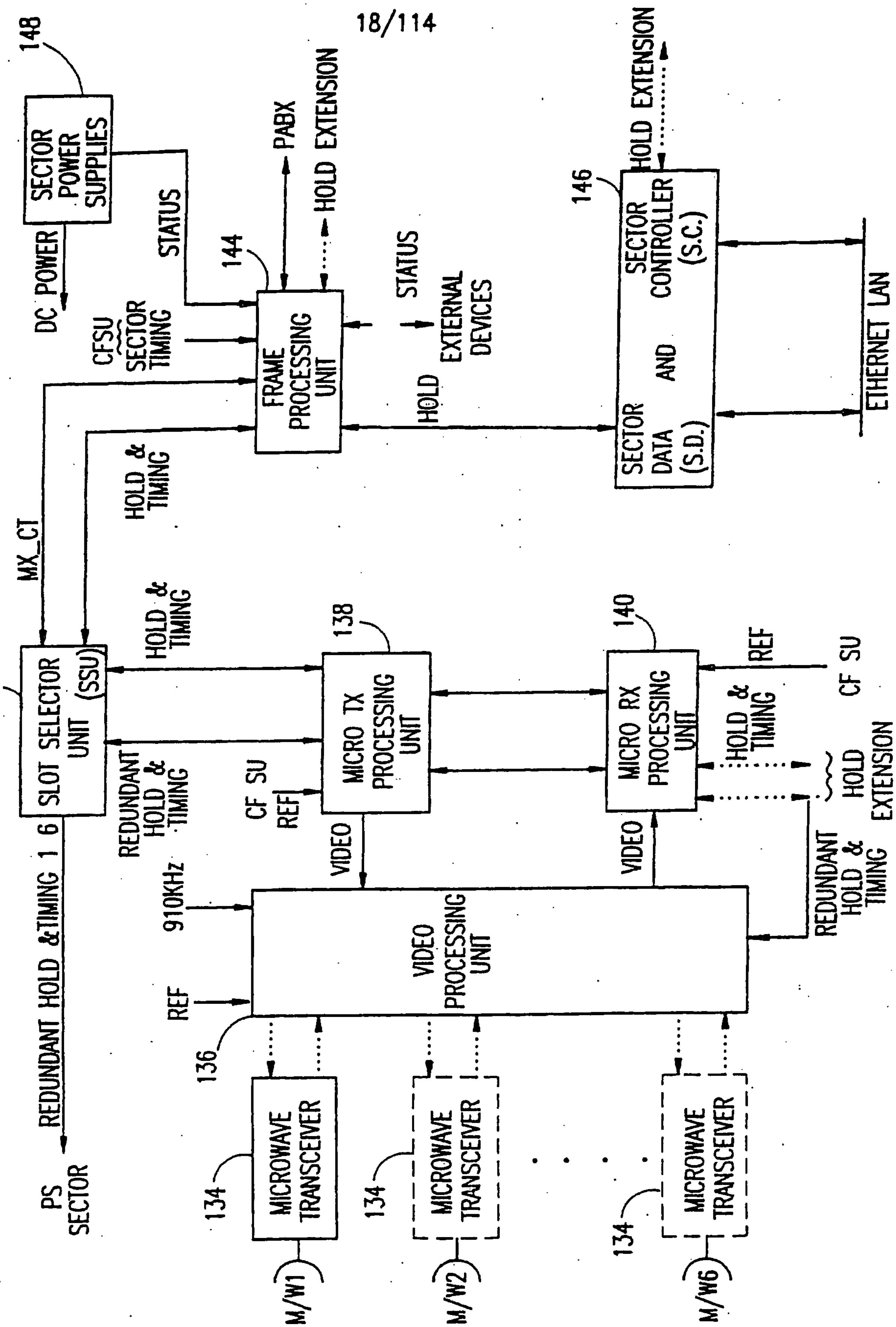

FRAME PROCESSING UNIT
REDUNDANT SECTOR
SB_B_M
MB_B_M
CB_B_M
SE_TM_M
SB_B_R
MB_B_R
CB_B_R
SE_TM_R
10M_REF IN
133
REF .OUT 1
REF .OUT 2
10M_REF_1
2
3
4
5
6
7
8
124
BRIDGE INTERFACE BOARD 2
MB_I_R
CB_I_R
SB_I_R
SE_TM_I_R
BRIDGE INTERFACE BOARD 1
122
CB_I_M
SB_I_M
SE_TM_I_M
MB_I_M
SLOT_NO
SLOT_NO
DUAL DIGITAL TX BOARD 1
DUAL DIGITAL TX BOARD 6
130
132
PAU BIT
PAU BIT
126
10M REF_1
10M REF_6
DUAL TX RF/IF BOARD 1
DUAL TX RF/IF BOARD 6
RX PROCESSING UNIT
70
RF_OUT_1
RF_OUT_2
A.M.1
RF_OUT_11
RF_OUT_12
128
A.M.3

FIG. 13B


148
SECTOR POWER SUPPLIES
DC POWER
STATUS
PABX
HOLD EXTENSION
144
CFSU SECTOR TIMING
FRAME PROCESSING UNIT
STATUS
EXTERNAL DEVICES
HOLD
146
HOLD EXTENSION
SECTOR CONTROLLER (S.C.)
AND
SECTOR DATA (S.D.)
ETHERNET LAN
MX_CT
HOLD & TIMING
HOLD & TIMING
142
SLOT SELECTOR UNIT (SSU)
138
MICRO TX PROCESSING UNIT
140
MICRO RX PROCESSING UNIT
REF
CF SU
HOLD & TIMING
HOLD EXTENSION
REDUNDANT HOLD & TIMING
CF SU
REF
VIDEO
VIDEO
REDUNDANT HOLD & TIMING 1 6
REDUNDANT HOLD & TIMING
910KHz
REF
VIDEO PROCESSING UNIT
136
134
MICROWAVE TRANSCEIVER
134
MICROWAVE TRANSCEIVER
134
MICROWAVE TRANSCEIVER
PS SECTOR
M/W1
M/W2
M/W6

34

FIG. 14

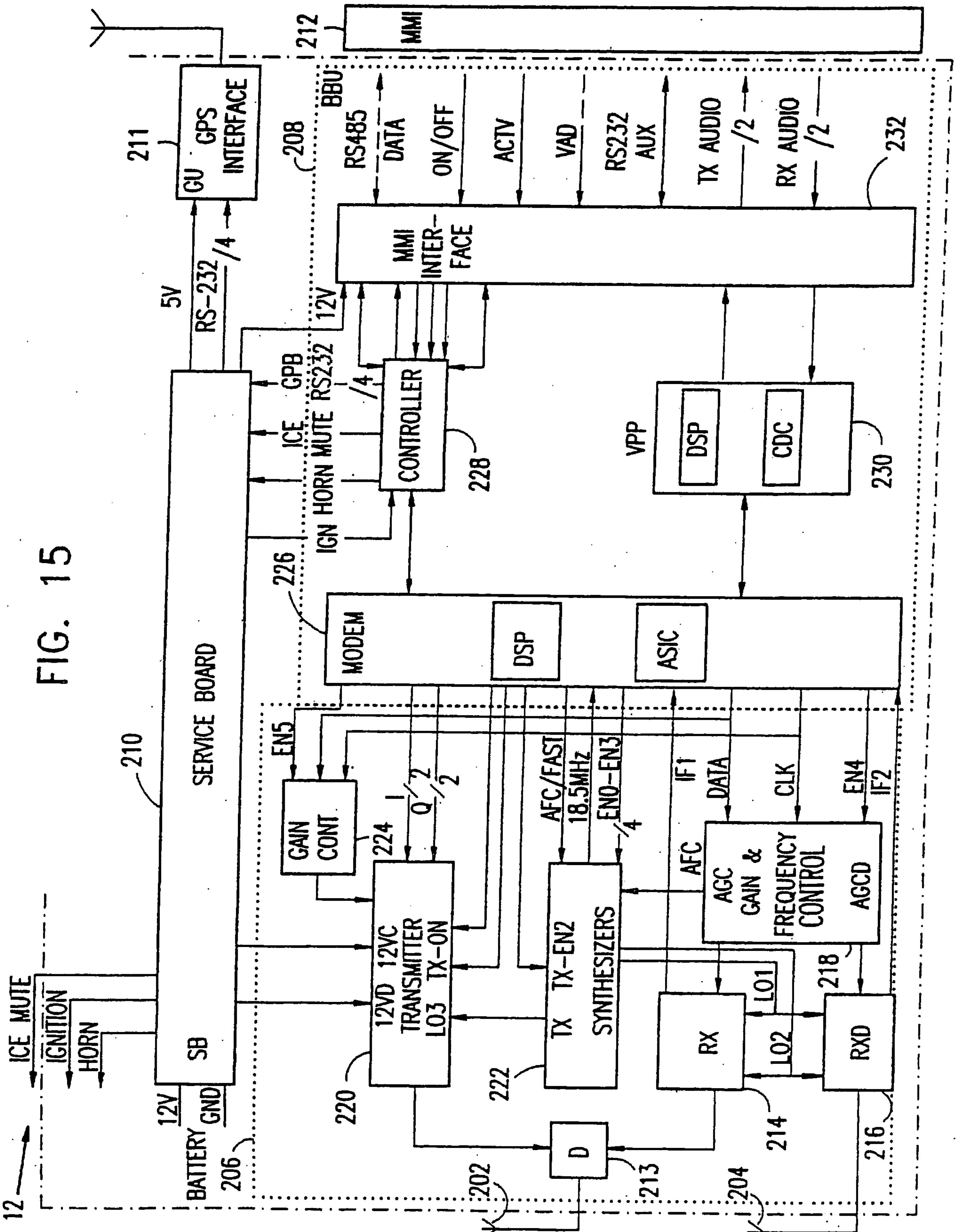

MMI
212
BBU
RS485
DATA
ON/OFF
ACTV
VAD
RS232
AUX
TX AUDIO
/2
RX AUDIO
/2
232
211
GU GPS INTERFACE
208
MMI INTER-FACE
5V
RS-232
/4
12V
GPB
RS232
/4
ICE
IGN HORN MUTE
CONTROLLER
228
VPP
DSP
CDC
230
226
MODEM
DSP
ASIC
SERVICE BOARD
210
SB
EN5
GAIN CONT
224
I
Q
/2
/2
AFC/FAST
18.5MHz
EN0-EN3
/4
IF1
DATA
CLK
EN4
IF2
AFC
AGC GAIN & FREQUENCY CONTROL
AGCD
12VD 12VC TRANSMITTER LO3 TX-ON
TX TX-EN2 SYNTHESIZERS
LO1
218
RX
LO2
RXD
220
222
214
216
ICE MUTE
IGNITION
HORN
12V
GND
BATTERY
206
D
213
202
204
12

FIG. 15

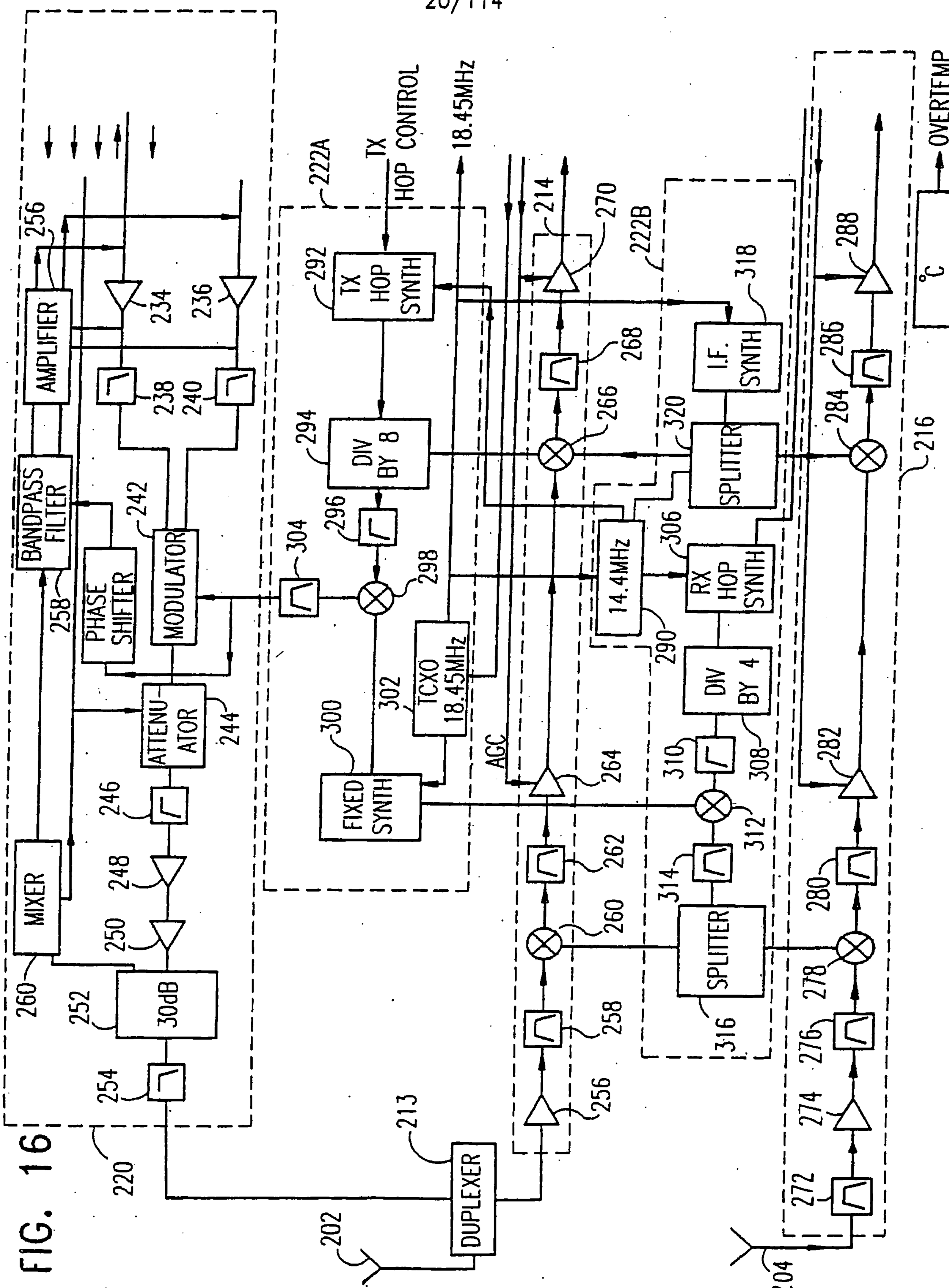


FIG. 16
AMPLIFIER
BANDPASS FILTER
PHASE SHIFTER
MODULATOR
ATTENUATOR
MIXER
30dB
DUPLEXER
TX HOP CONTROL
18.45MHz
TX HOP SYNTH
DIV BY 8
TCXO 18.45MHz
FIXED SYNTH
AGC
14.4MHz
RX HOP SYNTH
DIV BY 4
SPLITTER
I.F. SYNTH
°C
OVERTEMP
202
204
213
214
216
220
222A
222B
234
236
238
240
242
244
246
248
250
252
254
256
258
260
262
264
266
268
270
272
274
276
278
280
282
284
286
288
290
292
294
296
298
300
302
304
306
308
310
312
314
316
318
320

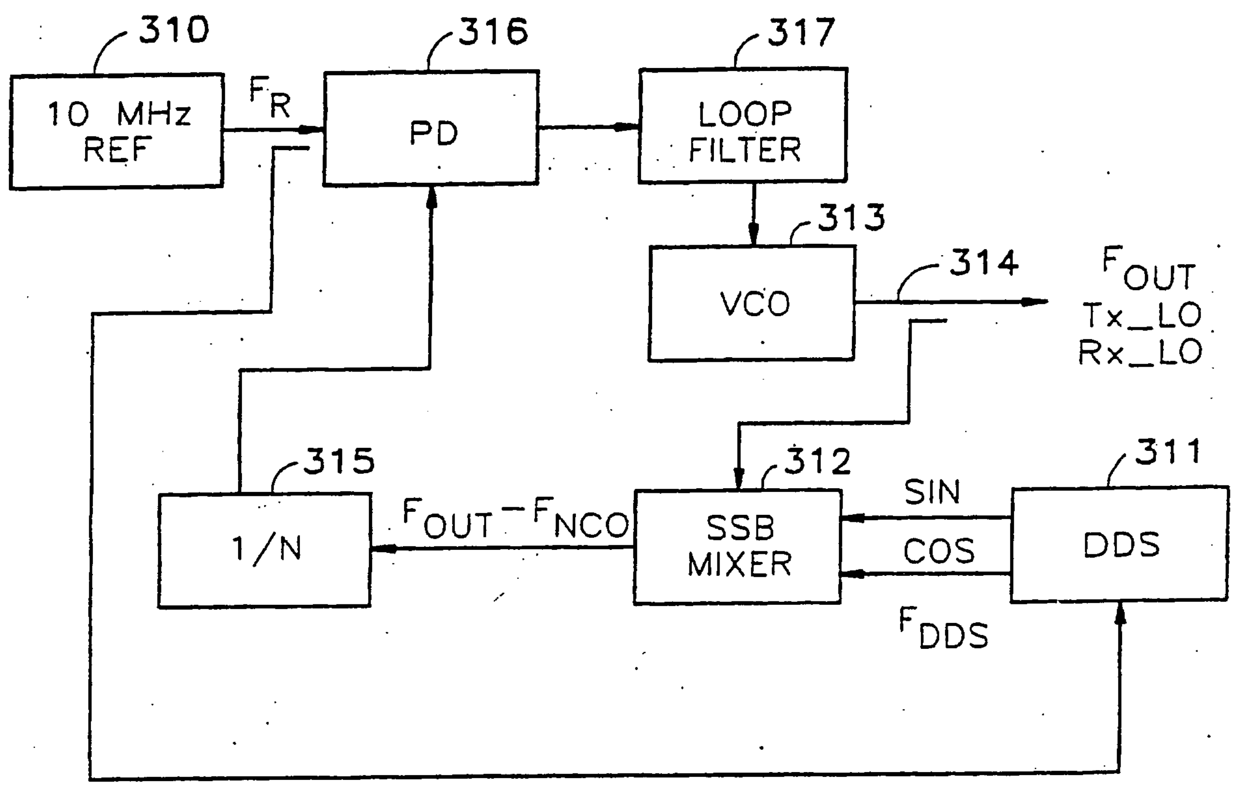
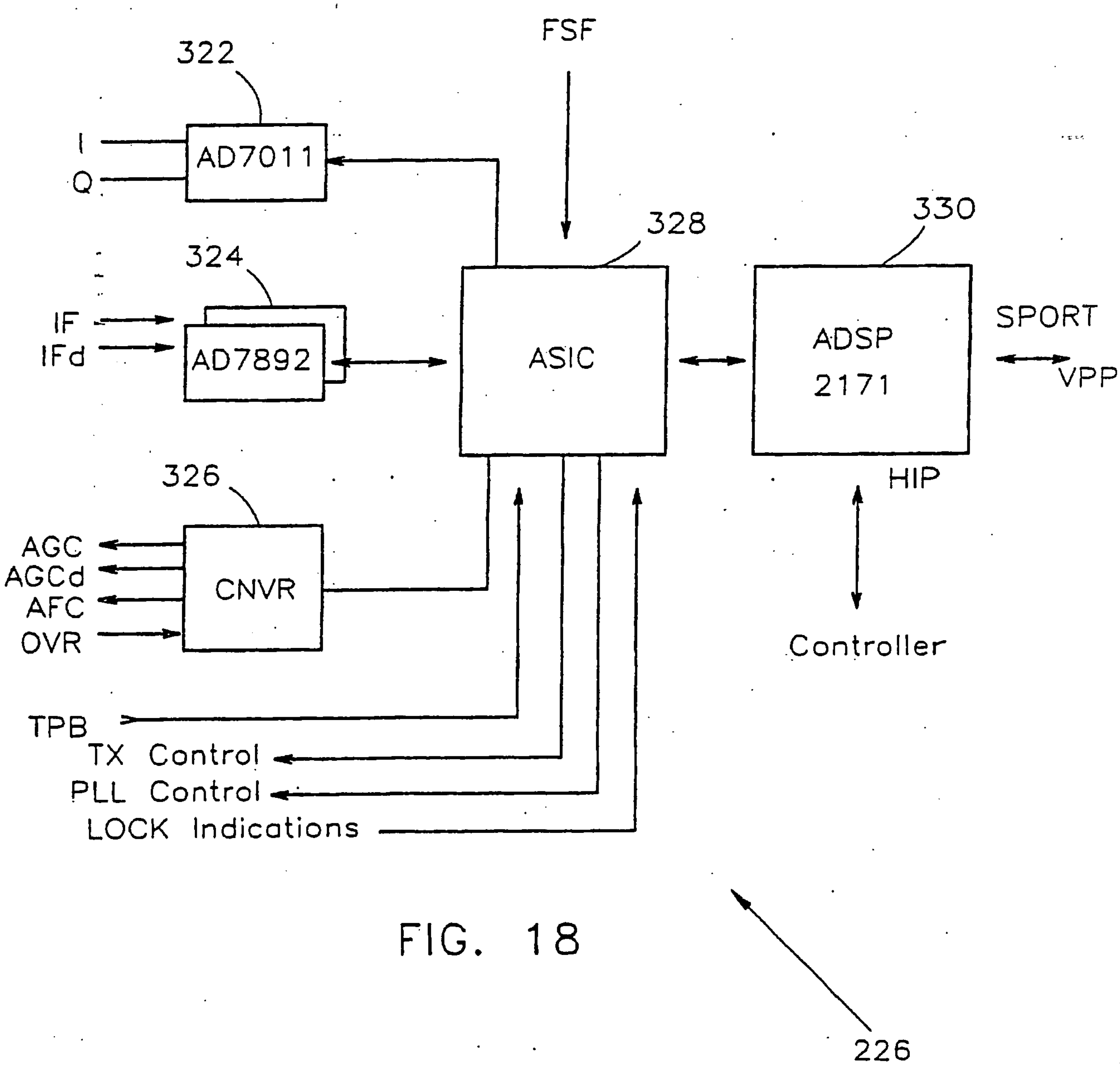

310
10 MHz
REF
F_R
316
PD
317
LOOP
FILTER
313
VCO
314
F_{OUT}
Tx_LO
Rx_LO
315
1/N
$F_{OUT}-F_{NCO}$
312
SSB
MIXER
SIN
COS
F_{DDS}
311
DDS

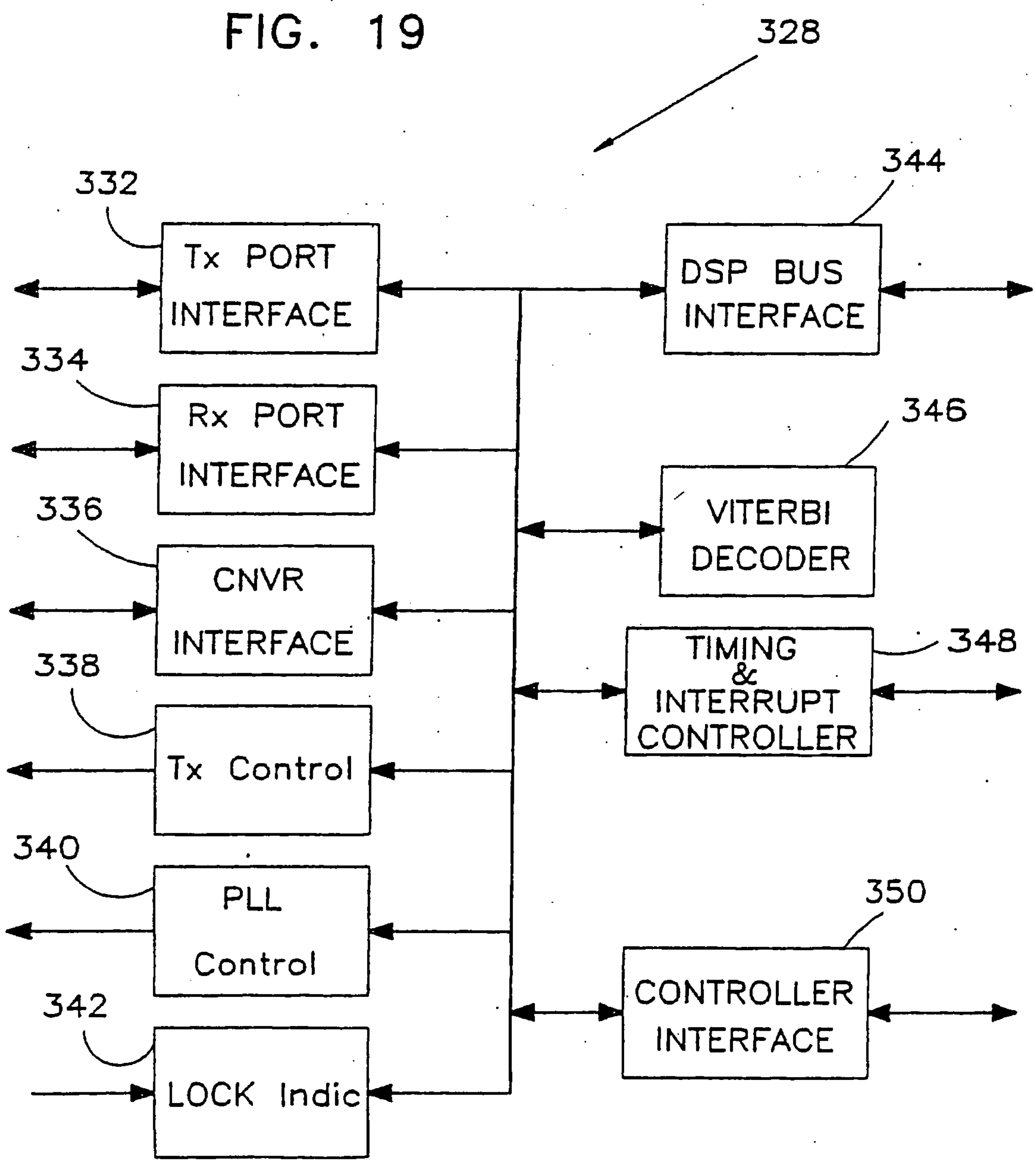
FIG. 17



FSF
322
I
Q
AD7011
328
330
324
IF
IFd
AD7892
ASIC
ADSP
2171
SPORT
VPP
326
HIP
AGC
AGCd
AFC
OVR
CNVR
Controller
TPB
TX Control
PLL Control
LOCK Indications
226

FIG. 18

FIG. 19


328
332
Tx PORT INTERFACE
334
Rx PORT INTERFACE
336
CNVR INTERFACE
338
Tx Control
340
PLL Control
342
LOCK Indic
344
DSP BUS INTERFACE
346
VITERBI DECODER
348
TIMING & INTERRUPT CONTROLLER
350
CONTROLLER INTERFACE

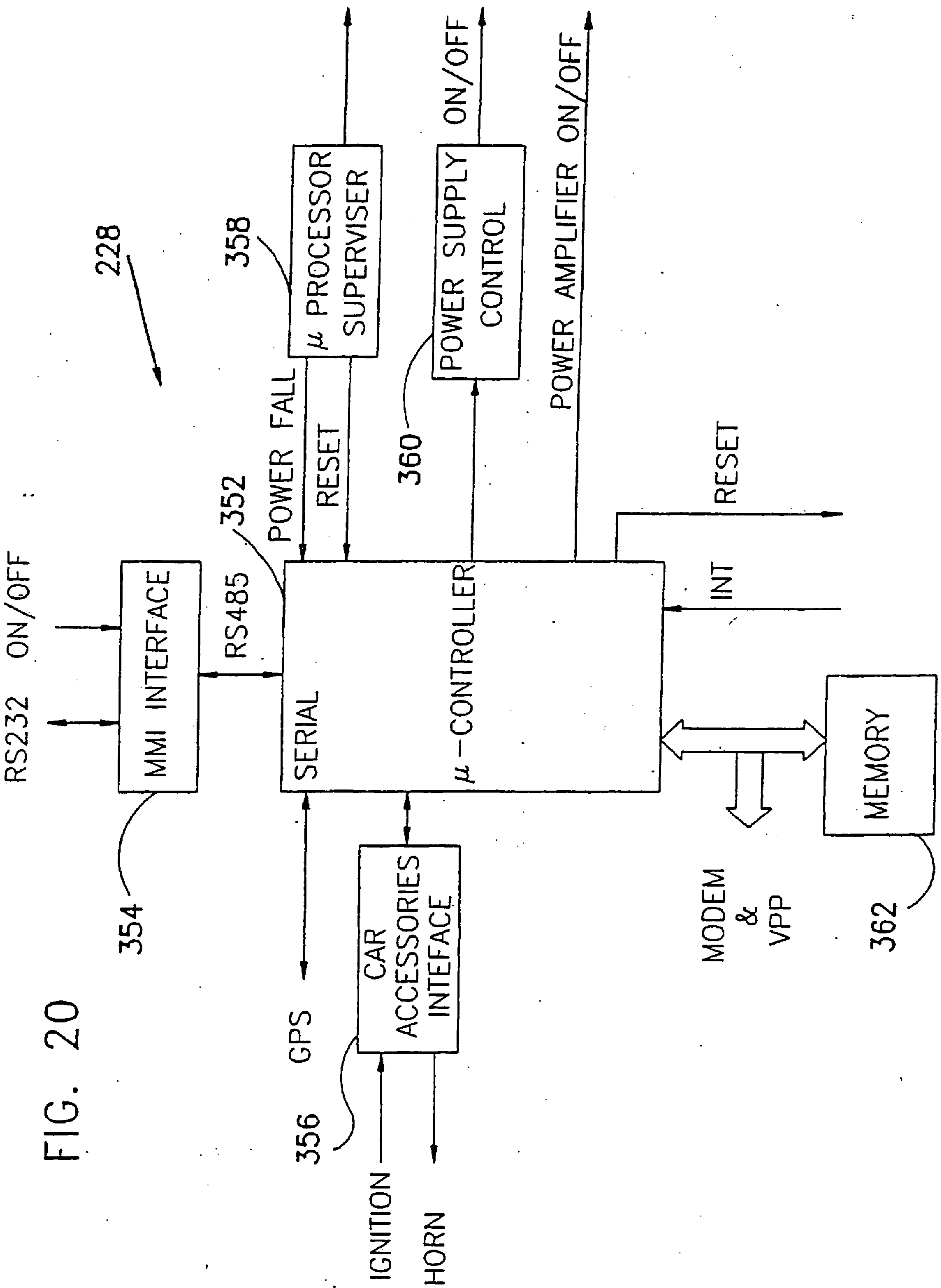

228
μ PROCESSOR SUPERVISER
358
POWER SUPPLY CONTROL
ON/OFF
360
POWER AMPLIFIER ON/OFF
POWER FALL
RESET
352
RESET
INT
ON/OFF
RS232
MMI INTERFACE
RS485
SERIAL
μ-CONTROLLER
MEMORY
354
MODEM & VPP
362
CAR ACCESSORIES INTEFACE
GPS
356
IGNITION
HORN

FIG. 20

FIG. 21

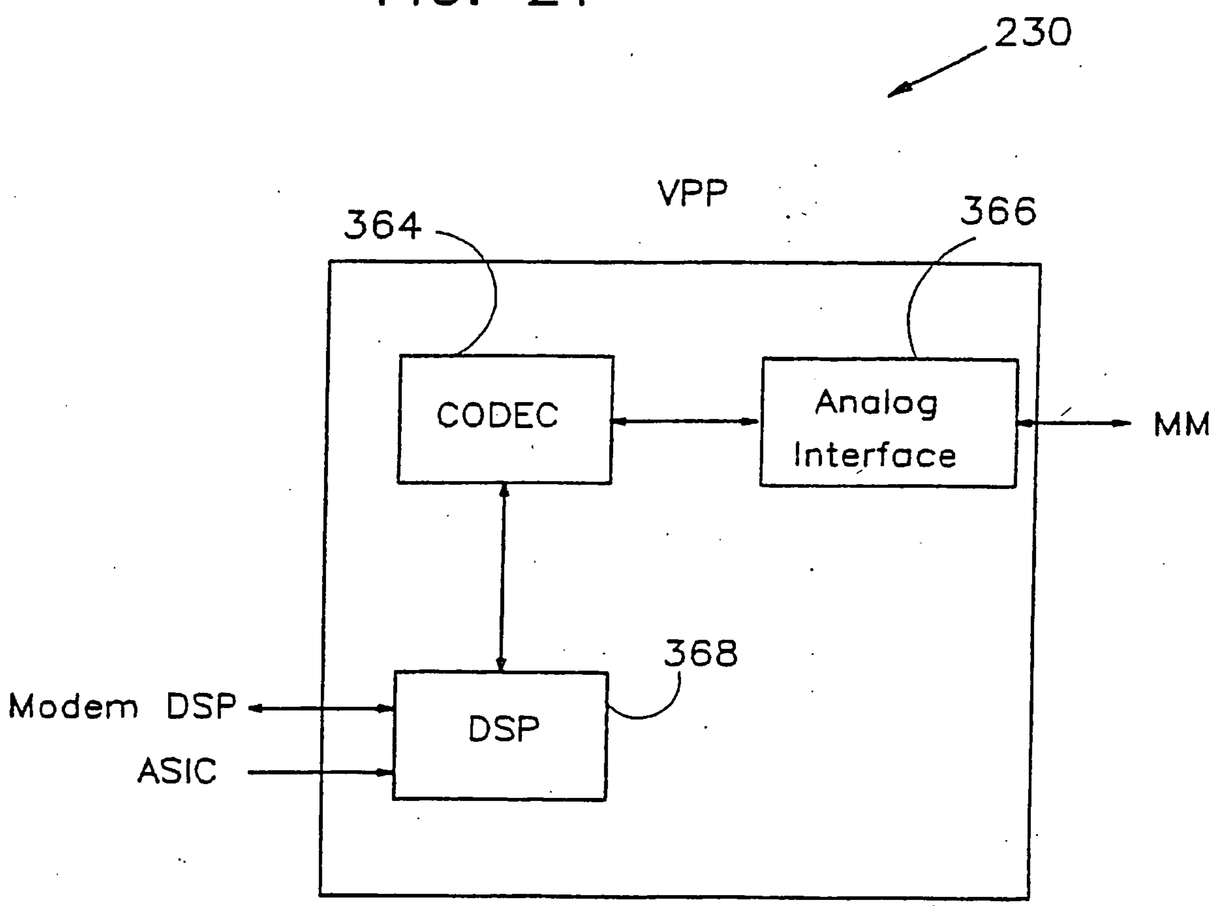
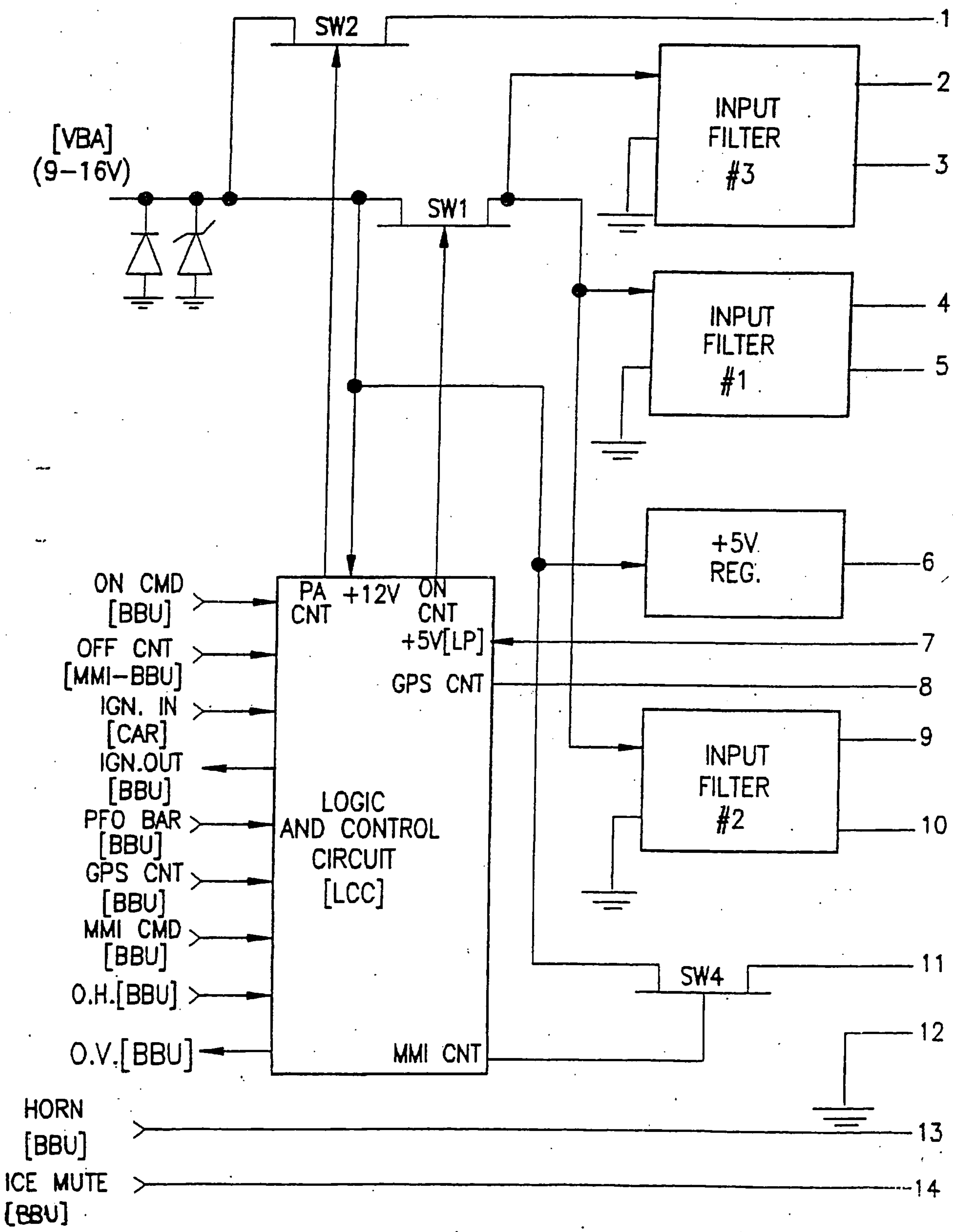

230
VPP
364
366
CODEC
Analog
Interface
MMI
368
Modem DSP
DSP
ASIC

FIG. 22A



SW2
1
INPUT FILTER #3
2
3
[VBA] (9–16V)
SW1
INPUT FILTER #1
4
5
+5V REG.
6
ON CMD [BBU]
PA CNT
+12V
ON CNT
+5V[LP]
7
OFF CNT [MMI–BBU]
GPS CNT
8
IGN. IN [CAR]
INPUT FILTER #2
9
IGN.OUT [BBU]
LOGIC AND CONTROL CIRCUIT [LCC]
PFO BAR [BBU]
10
GPS CNT [BBU]
MMI CMD [BBU]
SW4
11
O.H.[BBU]
12
O.V.[BBU]
MMI CNT
HORN [BBU]
13
ICE MUTE [BBU]
14

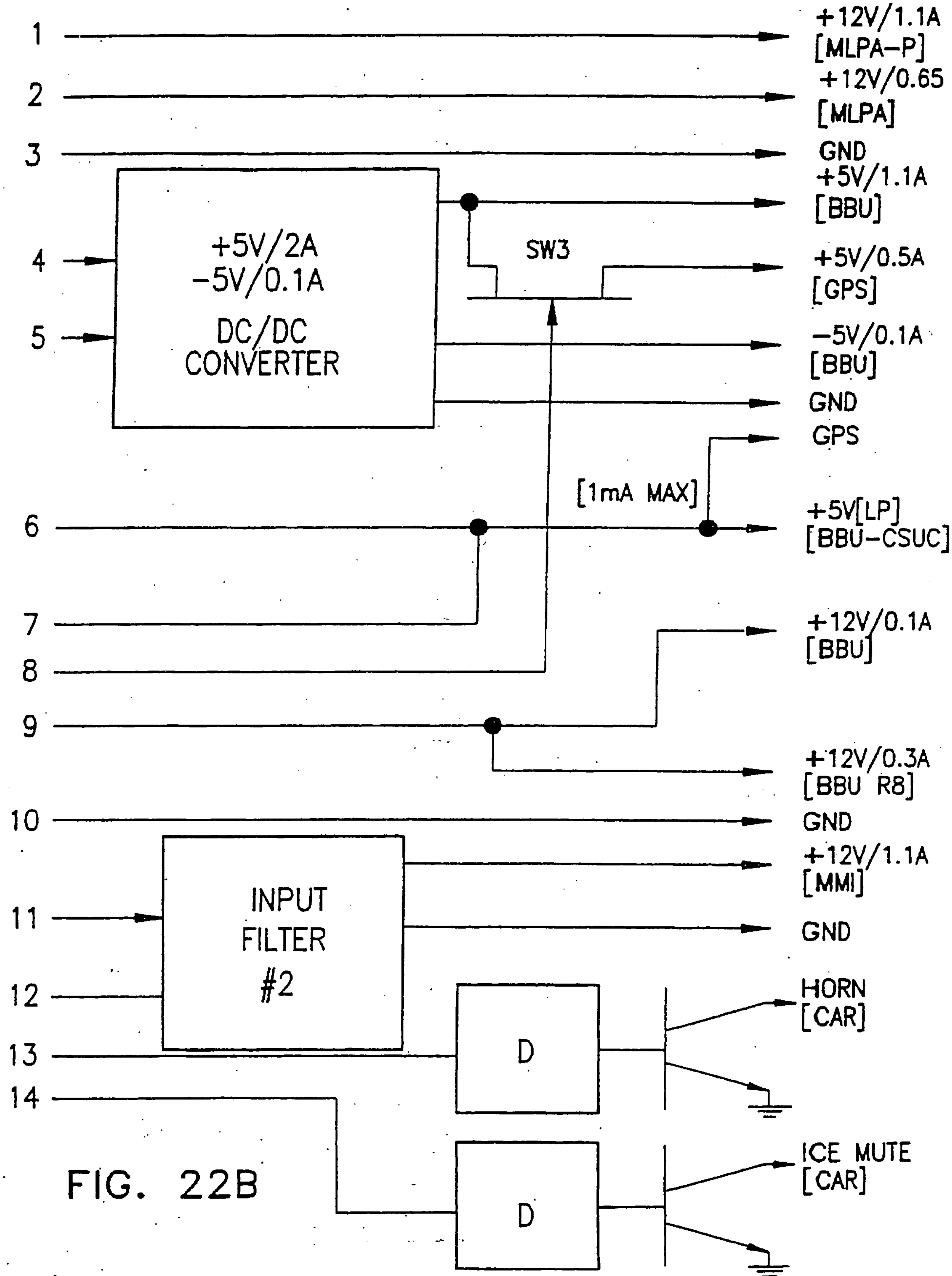
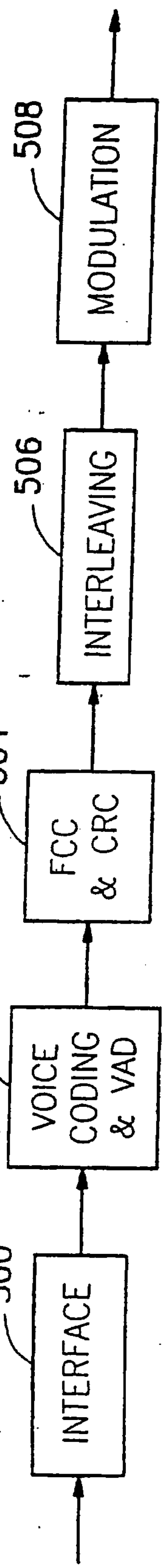
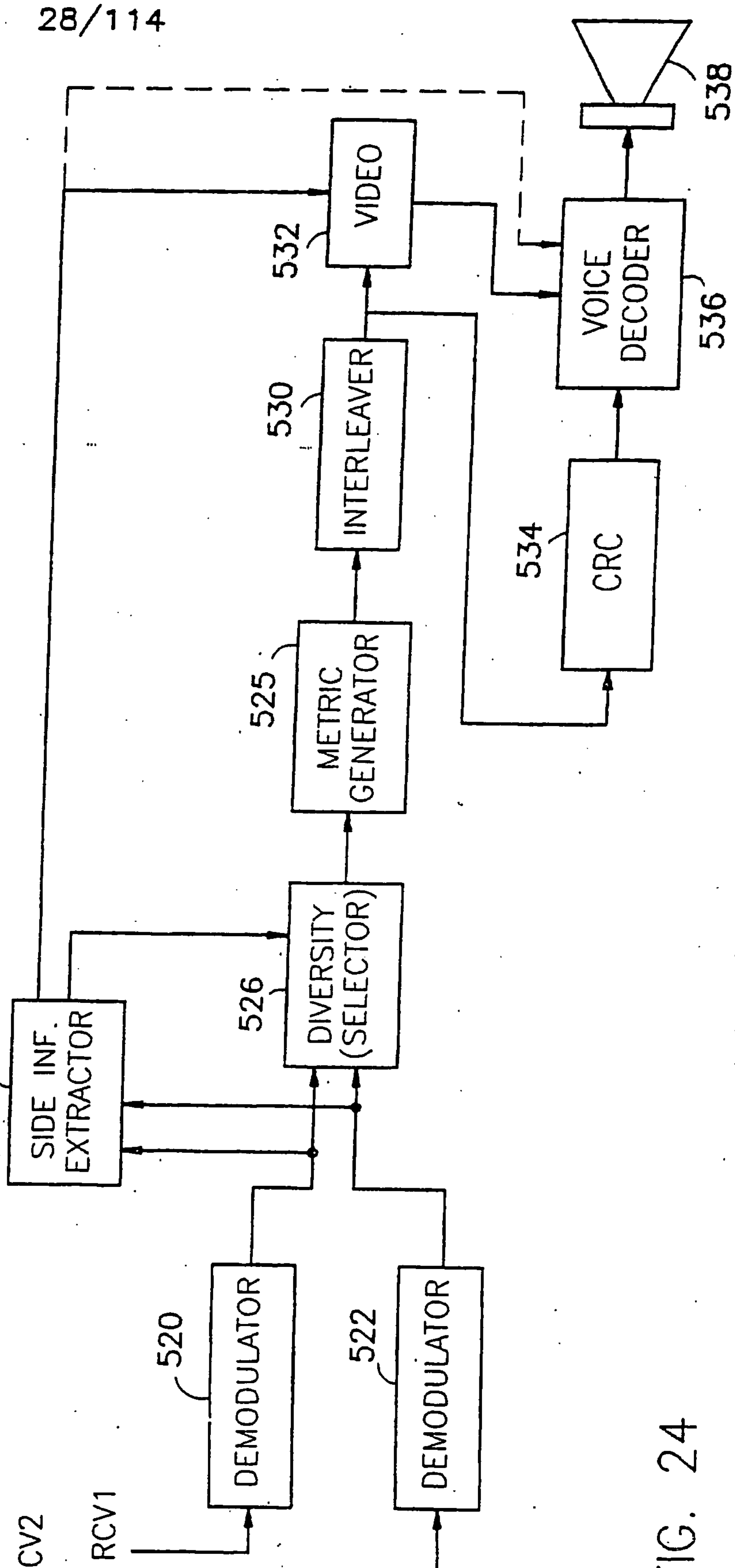

1
+12V/1.1A
[MLPA-P]
2
+12V/0.65
[MLPA]
3
GND
+5V/1.1A
[BBU]
+5V/2A
-5V/0.1A
DC/DC
CONVERTER
4
5
SW3
+5V/0.5A
[GPS]
-5V/0.1A
[BBU]
GND
GPS
[1mA MAX]
6
+5V[LP]
[BBU-CSUC]
7
8
+12V/0.1A
[BBU]
9
+12V/0.3A
[BBU R8]
10
GND
+12V/1.1A
[MMI]
INPUT
FILTER
#2
11
GND
12
13
D
HORN
[CAR]
14
D
ICE MUTE
[CAR]

FIG. 22B



INTERFACE
500
VOICE CODING & VAD
502
FCC & CRC
504
INTERLEAVING
506
MODULATION
508

FIG. 23



RCV1
CV2
DEMODULATOR
520
DEMODULATOR
522
SIDE INF. EXTRACTOR
524
DIVERSITY (SELECTOR)
526
METRIC GENERATOR
525
INTERLEAVER
530
VIDEO
532
CRC
534
VOICE DECODER
536
538

FIG. 24

FIG. 25

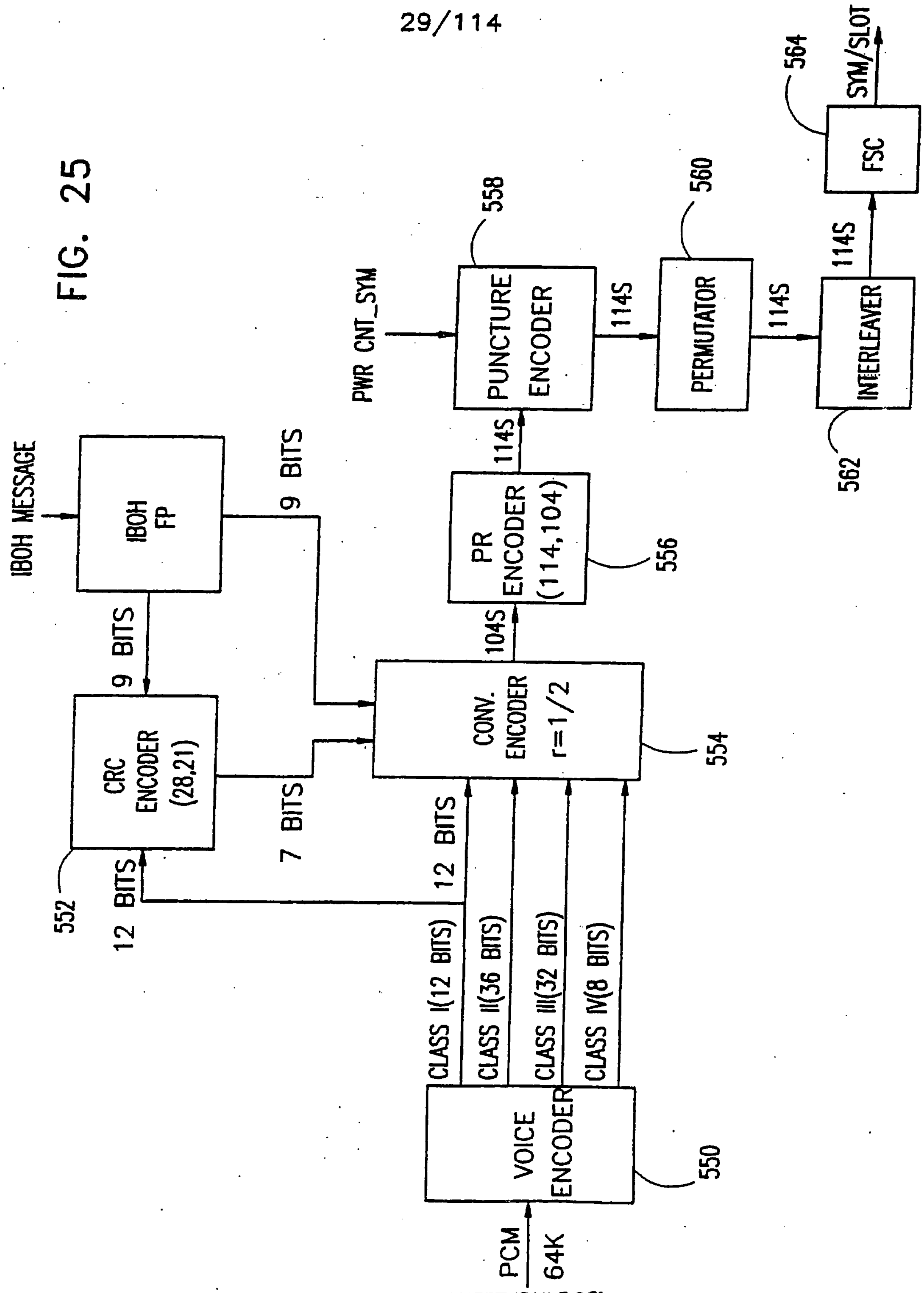
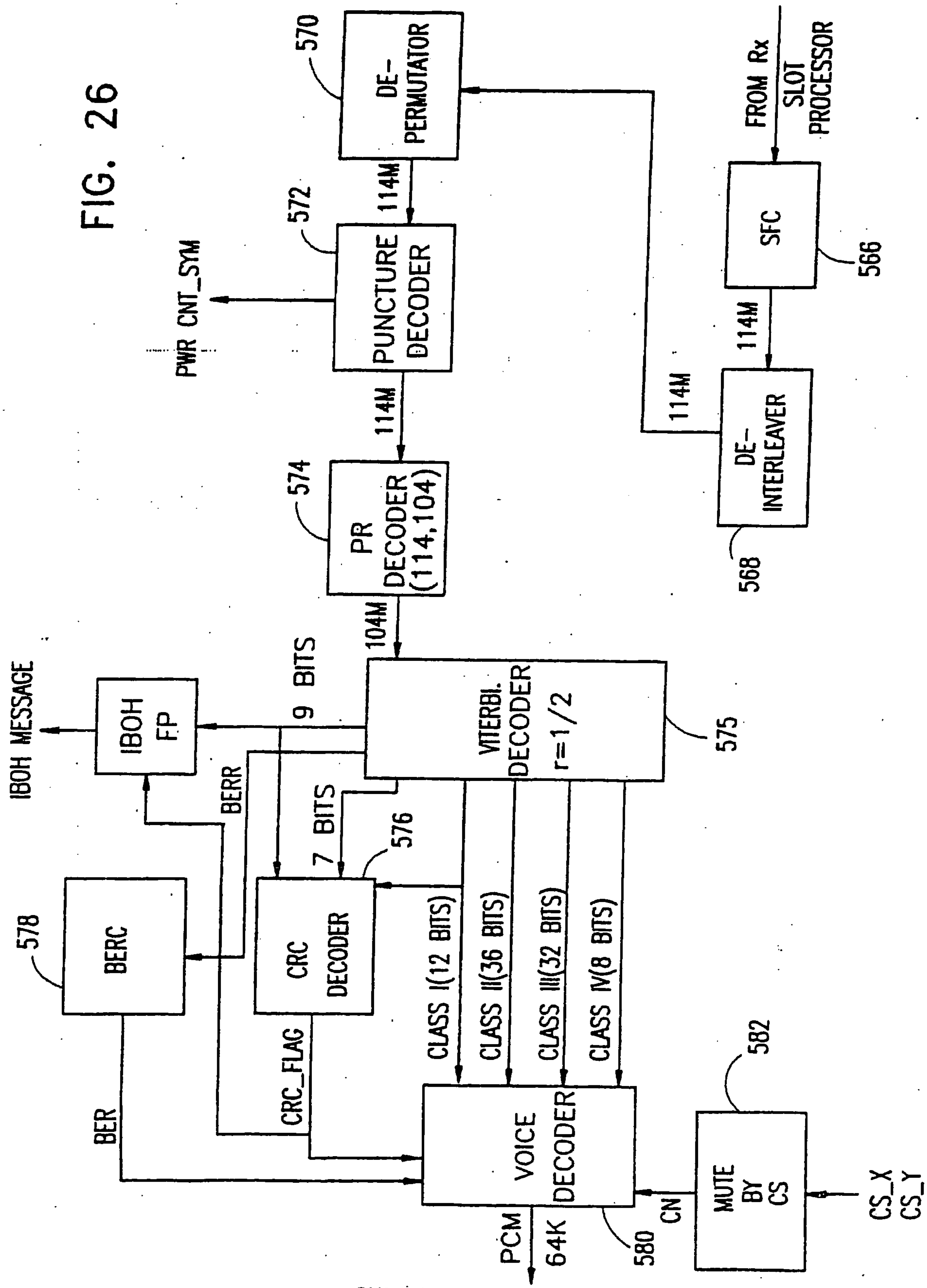

IBOH MESSAGE
IBOH FP
9 BITS
9 BITS
CRC ENCODER (28,21)
552
12 BITS
7 BITS
12 BITS
CONV. ENCODER r=1/2
554
104S
PR ENCODER (114,104)
556
114S
PWR CNT_SYM
PUNCTURE ENCODER
558
114S
PERMUTATOR
560
114S
INTERLEAVER
562
114S
FSC
564
SYM/SLOT
CLASS I(12 BITS)
CLASS II(36 BITS)
CLASS III(32 BITS)
CLASS IV(8 BITS)
VOICE ENCODER
550
PCM 64K

FIG. 26


DE-PERMUTATOR
570
114M
PUNCTURE DECODER
572
PWR CNT_SYM
114M
PR DECODER (114,104)
574
104M
VITERBI. DECODER r=1/2
575
9 BITS
7 BITS
IBOH FP
IBOH MESSAGE
BERR
CRC DECODER
576
BERC
578
CLASS I(12 BITS)
CLASS II(36 BITS)
CLASS III(32 BITS)
CLASS IV(8 BITS)
CRC_FLAG
BER
VOICE DECODER
580
PCM 64K
CN
MUTE BY CS
582
CS_X
CS_Y
FROM Rx SLOT PROCESSOR
SFC
566
114M
DE-INTERLEAVER
568
114M

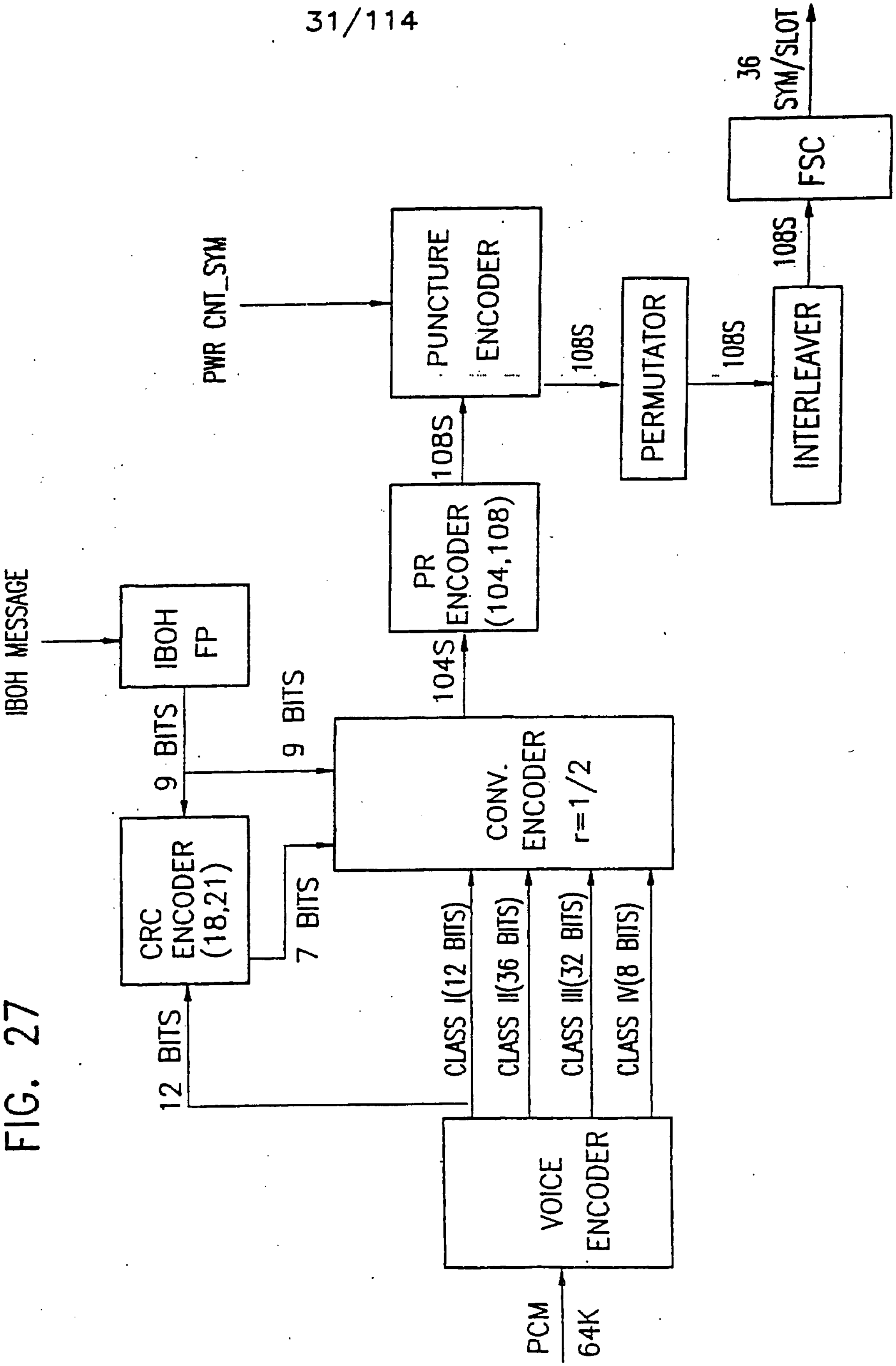
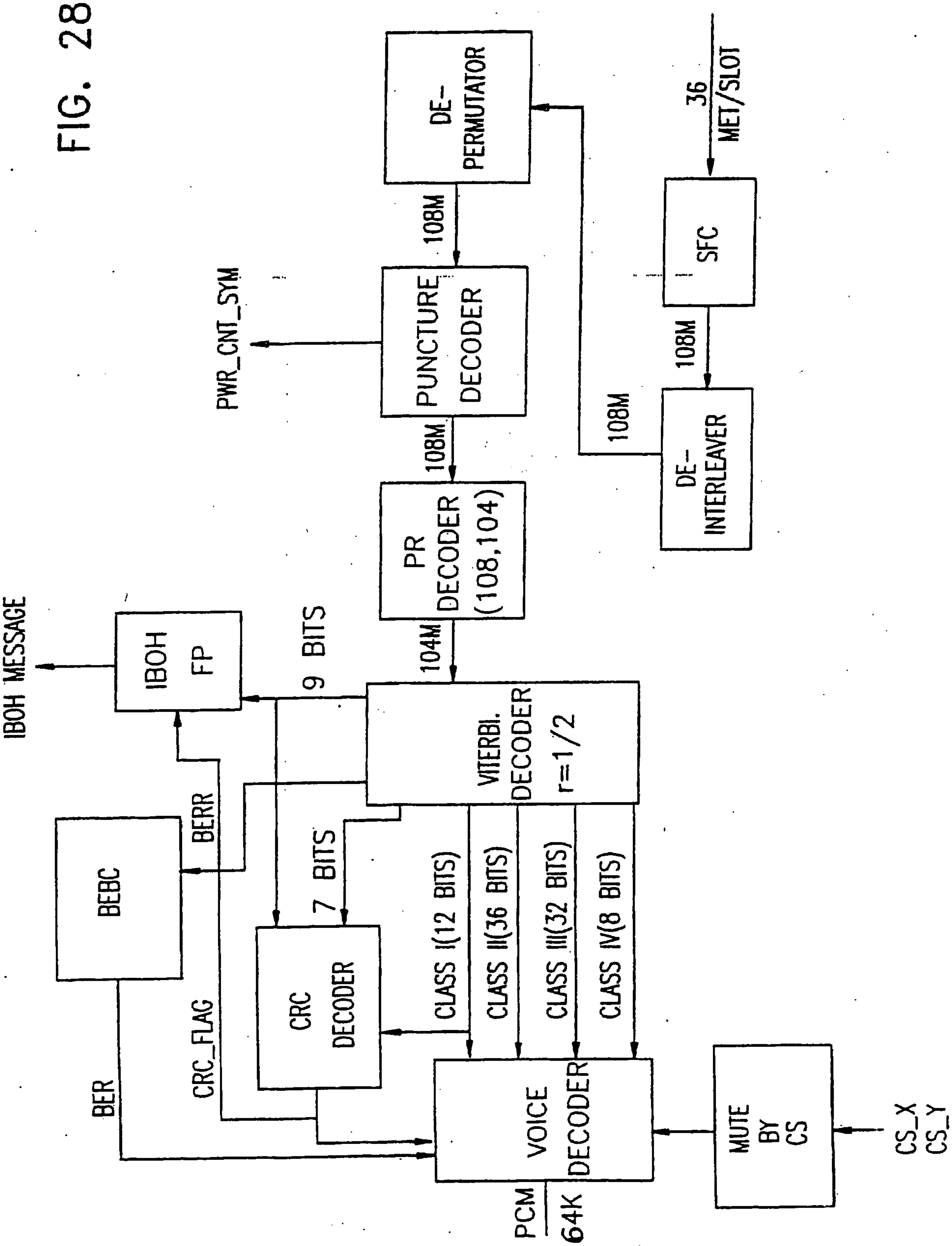

IBOH MESSAGE
IBOH FP
9 BITS
9 BITS
CRC ENCODER (18,21)
12 BITS
7 BITS
VOICE ENCODER
PCM 64K
CLASS I(12 BITS)
CLASS II(36 BITS)
CLASS III(32 BITS)
CLASS IV(8 BITS)
CONV. ENCODER r=1/2
104S
PR ENCODER (104,108)
108S
PWR CNT_SYM
PUNCTURE ENCODER
108S
PERMUTATOR
108S
INTERLEAVER
108S
FSC
36 SYM/SLOT

FIG. 27

FIG. 28


DE-
PERMUTATOR
36
MET/SLOT
108M
SFC
PWR_CNT_SYM
PUNCTURE
DECODER
108M
108M
DE-
INTERLEAVER
108M
PR
DECODER
(108,104)
IBOH MESSAGE
IBOH
FP
9 BITS
104M
VITERBI.
DECODER
r=1/2
BEBC
BERR
7 BITS
CRC
DECODER
CLASS I(12 BITS)
CLASS II(36 BITS)
CLASS III(32 BITS)
CLASS IV(8 BITS)
CRC_FLAG
BER
VOICE
DECODER
MUTE
BY
CS
CS_X
CS_Y
PCM
64K

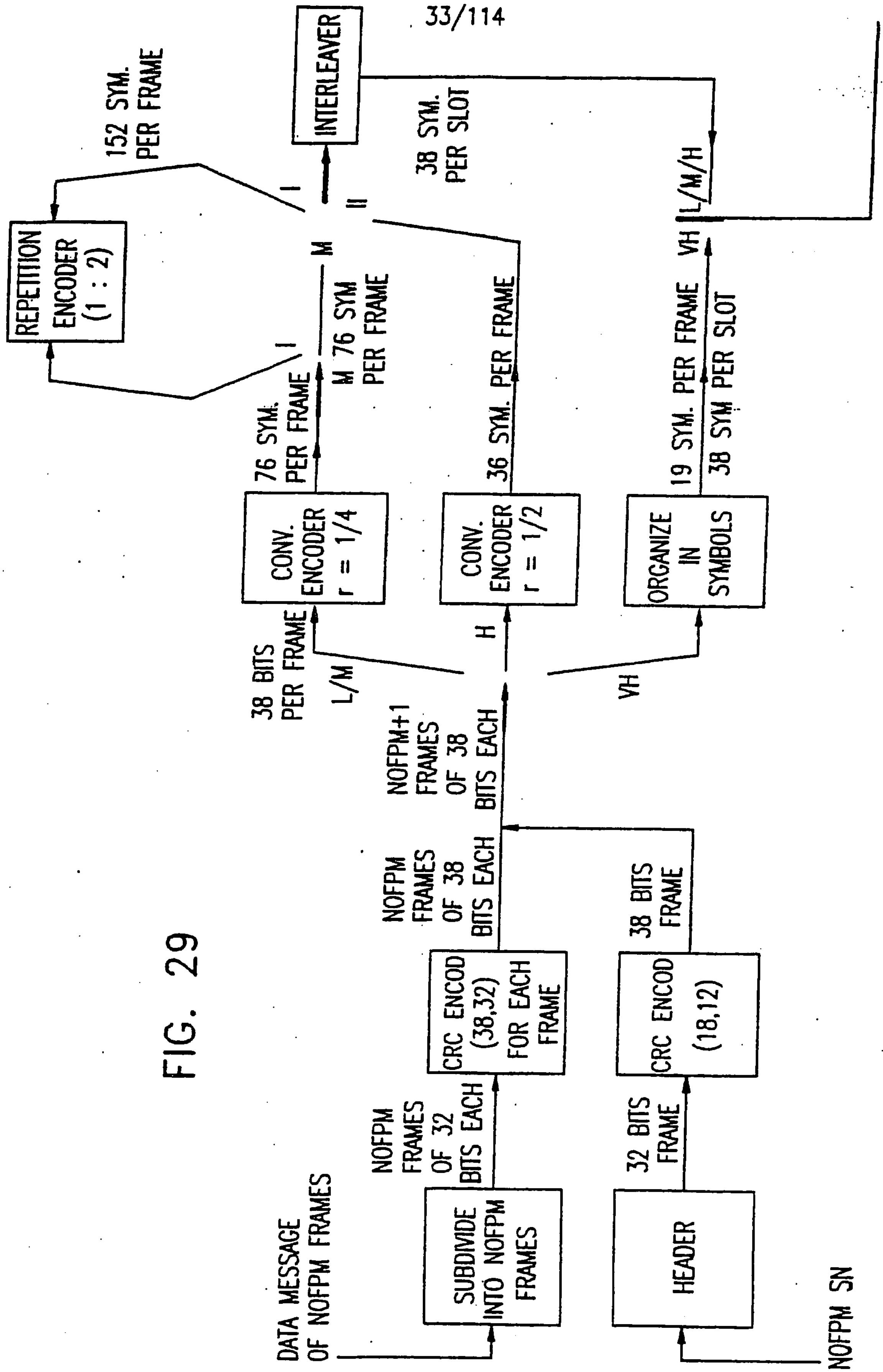

DATA MESSAGE OF NOFPM FRAMES
SUBDIVIDE INTO NOFPM FRAMES
NOFPM FRAMES OF 32 BITS EACH
CRC ENCOD (38,32) FOR EACH FRAME
NOFPM FRAMES OF 38 BITS EACH
NOFPM SN
HEADER
32 BITS FRAME
CRC ENCOD (18,12)
38 BITS FRAME
NOFPM+1 FRAMES OF 38 BITS EACH
L/M
H
VH
38 BITS PER FRAME
CONV. ENCODER r = 1/4
CONV. ENCODER r = 1/2
ORGANIZE IN SYMBOLS
76 SYM. PER FRAME
36 SYM. PER FRAME
19 SYM. PER FRAME
38 SYM PER SLOT
M 76 SYM PER FRAME
L
M
REPETITION ENCODER (1 : 2)
152 SYM. PER FRAME
INTERLEAVER
38 SYM. PER SLOT
L/M/H
VH

FIG. 29

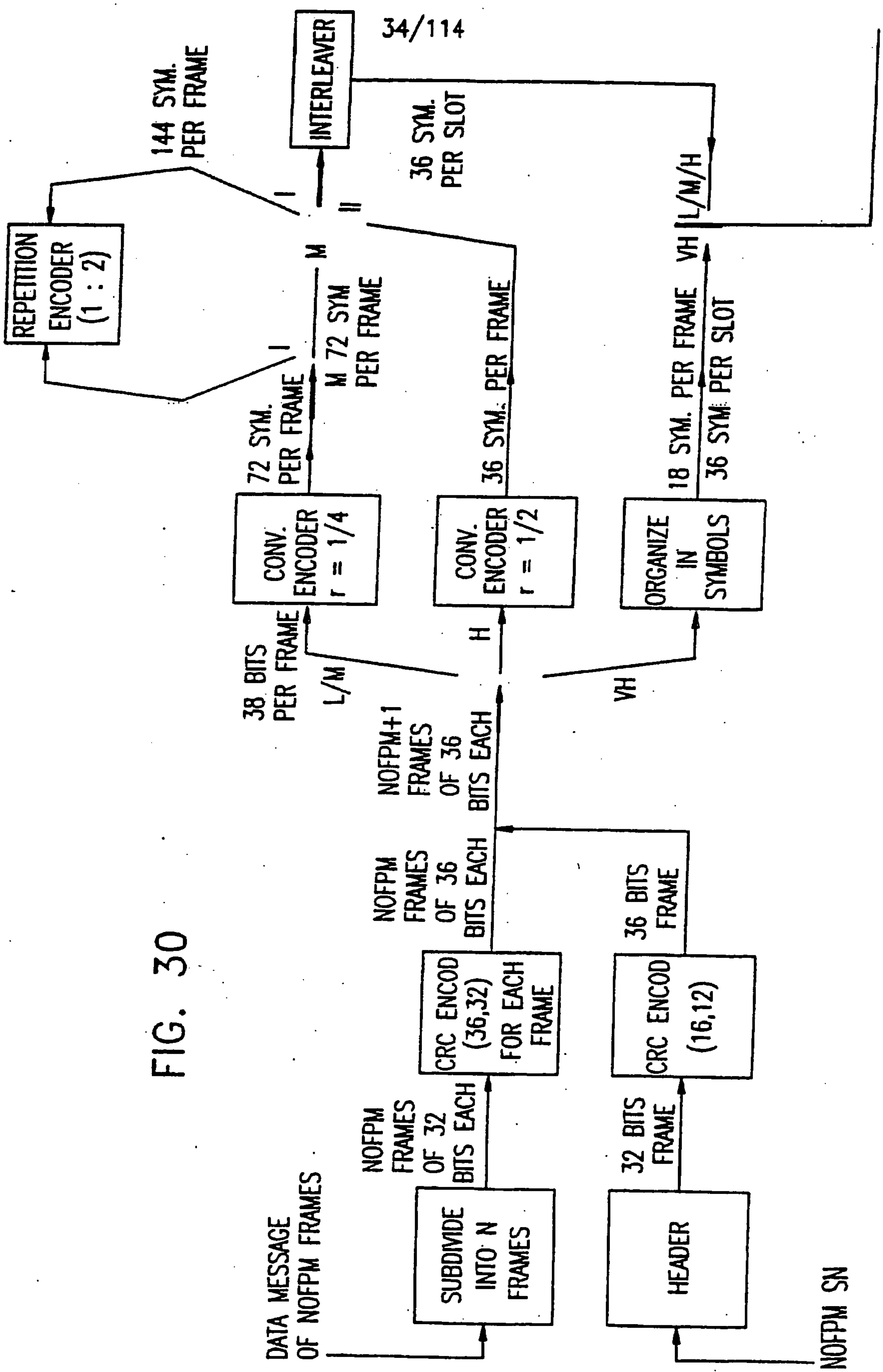
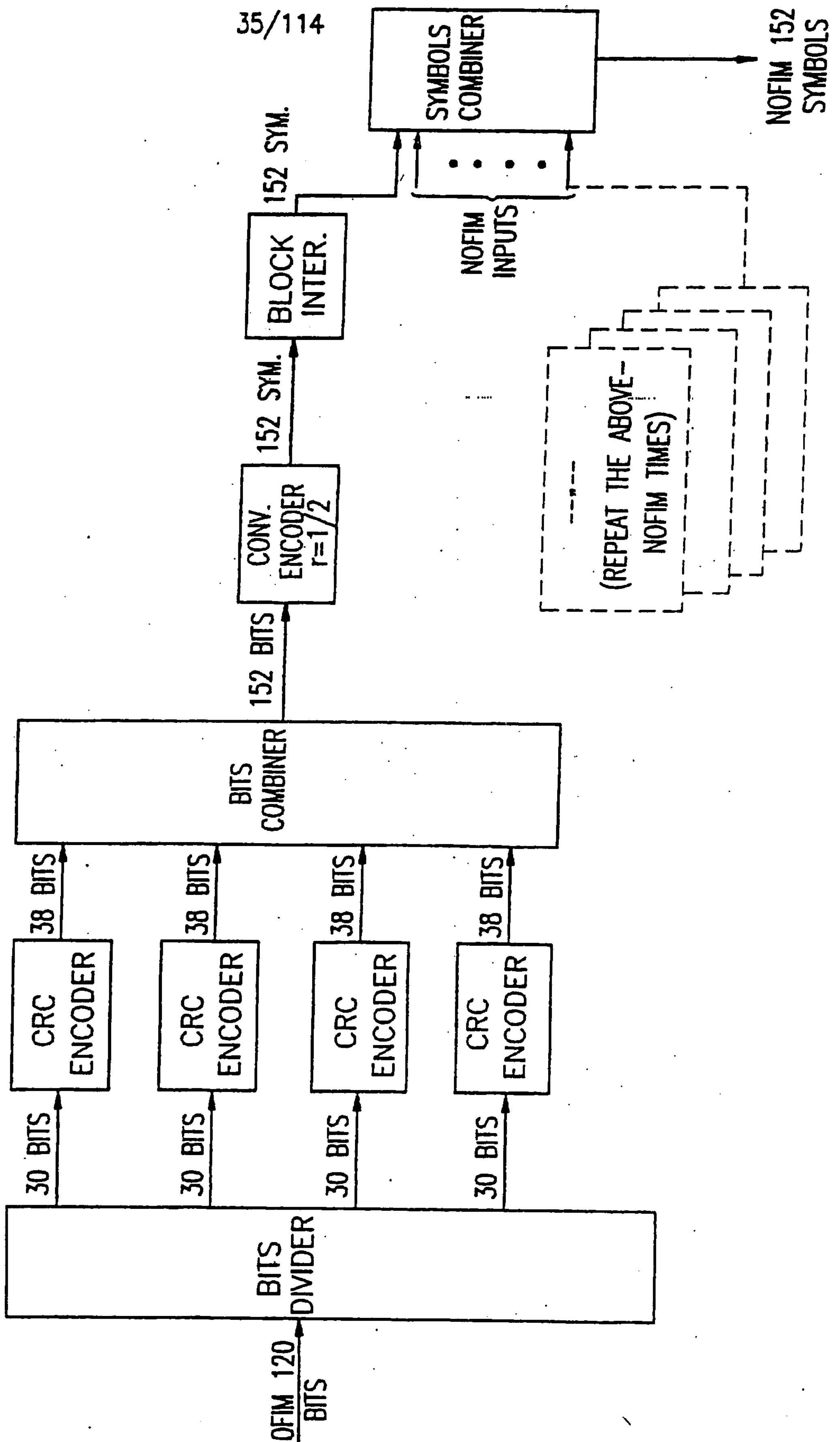

DATA MESSAGE OF NOFPM FRAMES
SUBDIVIDE INTO N FRAMES
NOFPM FRAMES OF 32 BITS EACH
CRC ENCOD (36,32) FOR EACH FRAME
NOFPM FRAMES OF 36 BITS EACH
NOFPM SN
HEADER
32 BITS FRAME
CRC ENCOD (16,12)
36 BITS FRAME
NOFPM+1 FRAMES OF 36 BITS EACH
L/M
H
VH
38 BITS PER FRAME
CONV. ENCODER r = 1/4
CONV. ENCODER r = 1/2
ORGANIZE IN SYMBOLS
72 SYM. PER FRAME
36 SYM. PER FRAME
18 SYM. PER FRAME
36 SYM. PER SLOT
M 72 SYM PER FRAME
L
M
REPETITION ENCODER (1 : 2)
144 SYM. PER FRAME
INTERLEAVER
36 SYM. PER SLOT
L/M/H
VH

FIG. 30

FIG. 31


NOFIM 120 BITS
BITS DIVIDER
30 BITS
30 BITS
30 BITS
30 BITS
CRC ENCODER
CRC ENCODER
CRC ENCODER
CRC ENCODER
38 BITS
38 BITS
38 BITS
38 BITS
BITS COMBINER
152 BITS
CONV. ENCODER r=1/2
152 SYM.
BLOCK INTER.
152 SYM.
SYMBOLS COMBINER
NOFIM 152 SYMBOLS
NOFIM INPUTS
(REPEAT THE ABOVE-NOFIM TIMES)

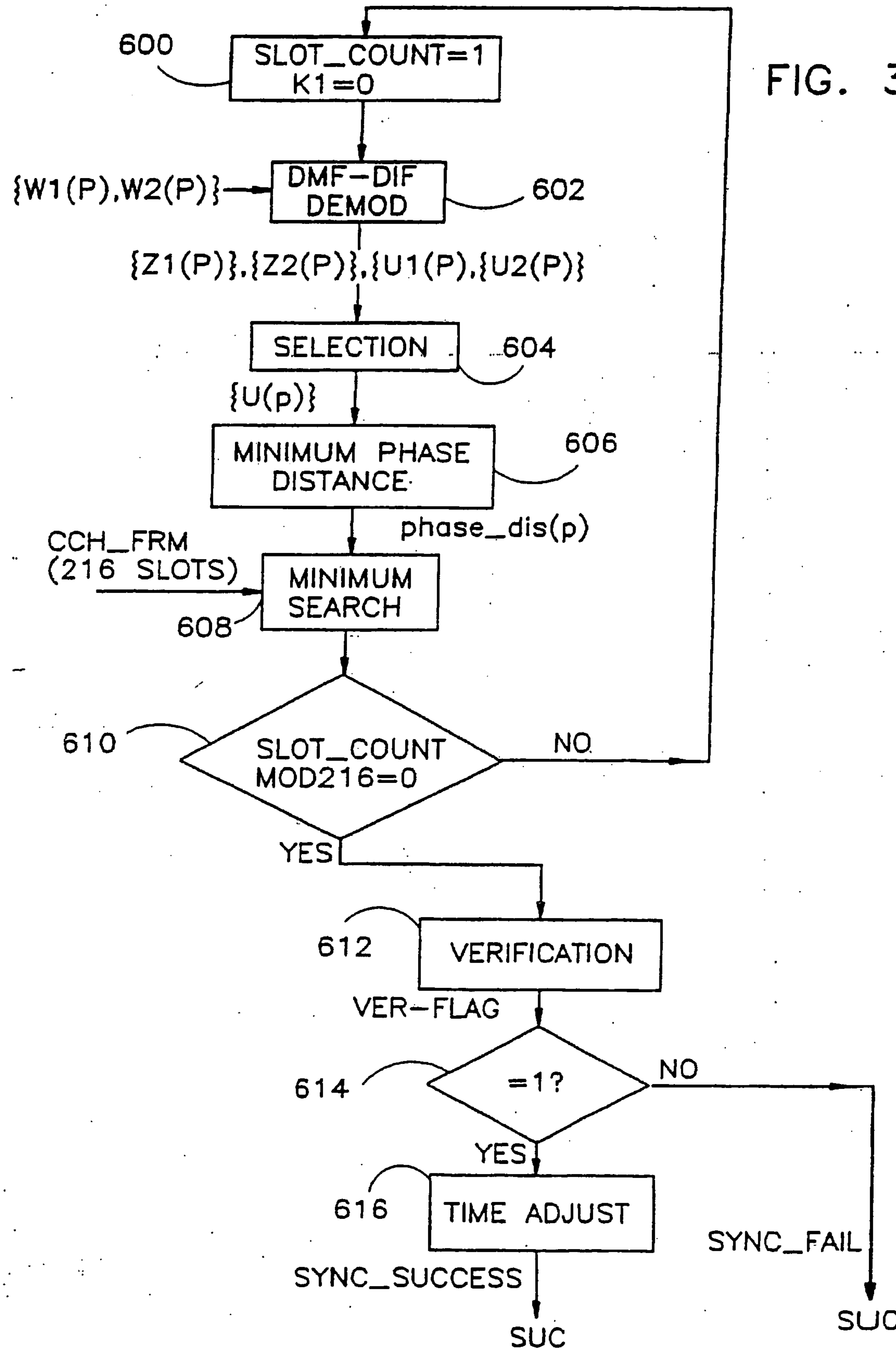

600
SLOT_COUNT=1
K1=0
{W1(P),W2(P)}
DMF-DIF
DEMOD
602
{Z1(P)},{Z2(P)},{U1(P),{U2(P)}
SELECTION
604
{U(p)}
MINIMUM PHASE
DISTANCE
606
phase_dis(p)
CCH_FRM
(216 SLOTS)
MINIMUM
SEARCH
608
610
SLOT_COUNT
MOD216=0
NO
YES
612
VERIFICATION
VER-FLAG
614
=1?
NO
YES
616
TIME ADJUST
SYNC_FAIL
SYNC_SUCCESS
SUC
SUC

FIG. 32

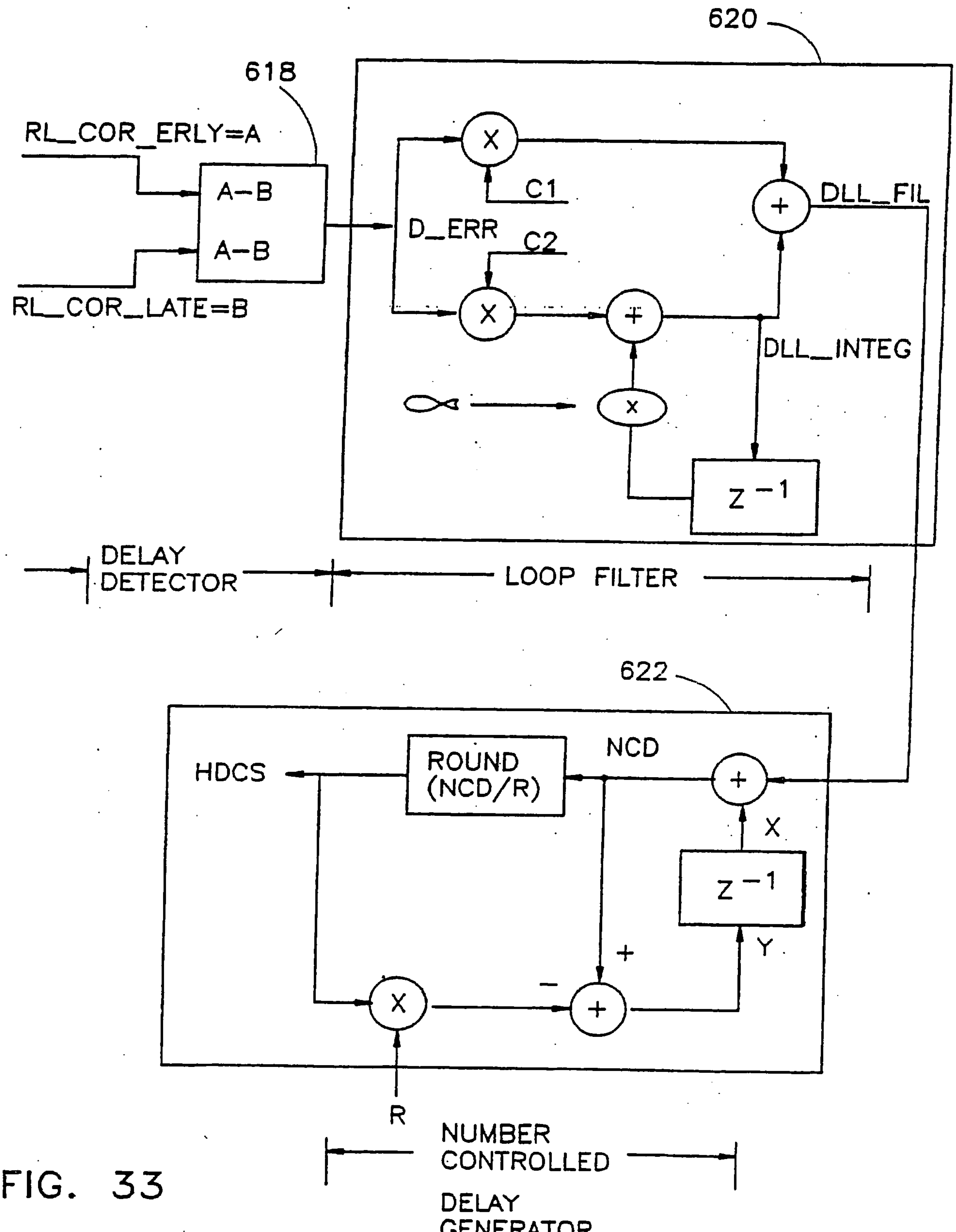

620
618
RL_COR_ERLY=A
A–B
A–B
RL_COR_LATE=B
D_ERR
X
C1
C2
X
+
+
DLL_FIL
DLL_INTEG
x
Z −1
DELAY DETECTOR
LOOP FILTER
622
HDCS
ROUND (NCD/R)
NCD
+
X
Z −1
Y
+
−
X
+
R
NUMBER CONTROLLED DELAY GENERATOR

FIG. 33

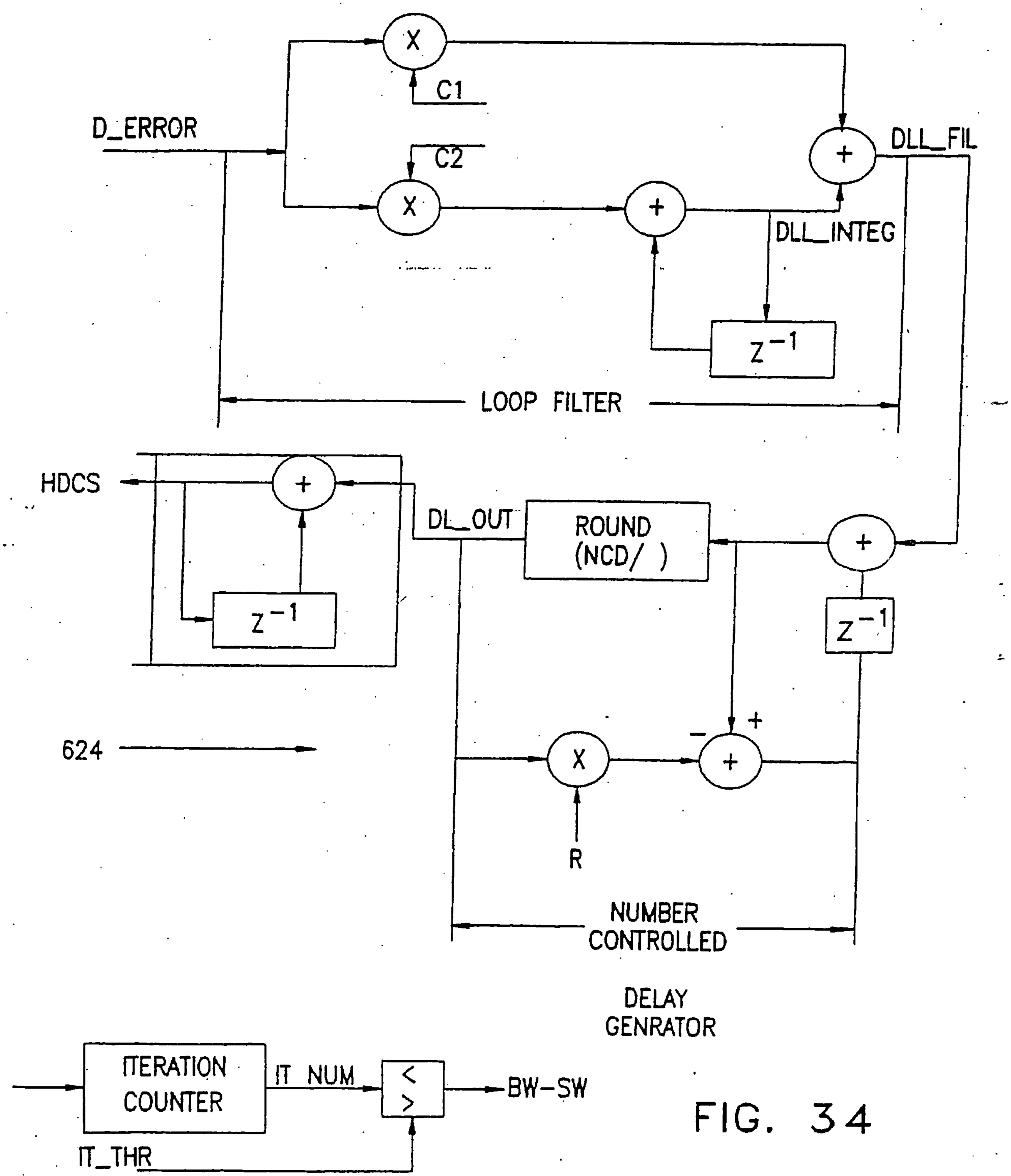
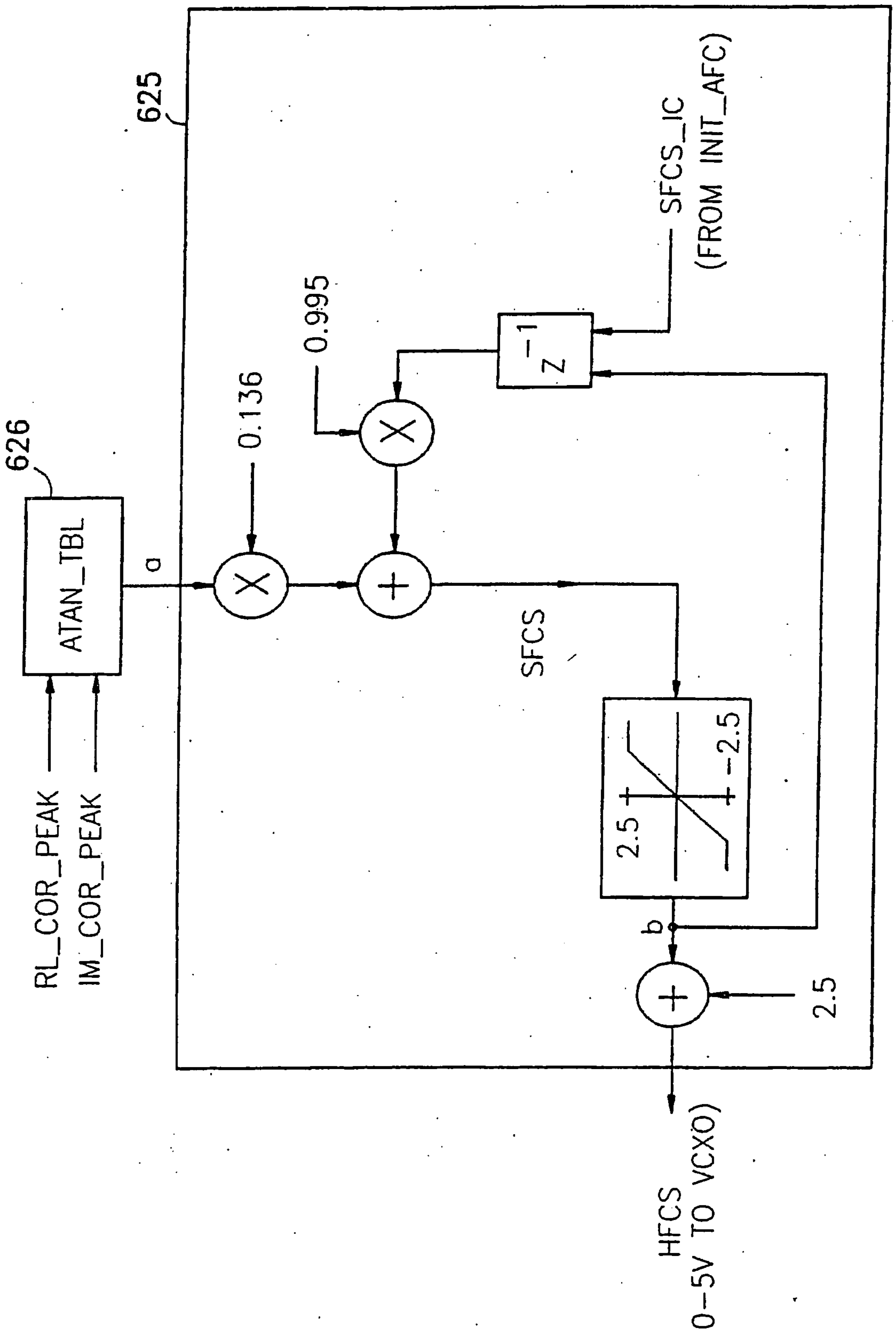

D_ERROR
X
C1
C2
X
+
+
DLL_FIL
DLL_INTEG
Z^{-1}
LOOP FILTER
HDCS
+
Z^{-1}
DL_OUT
ROUND (NCD/)
+
Z^{-1}
624
X
R
−
+
+
NUMBER CONTROLLED
DELAY GENRATOR
ITERATION COUNTER
IT NUM
IT_THR
<
>
BW-SW

FIG. 34



625
626
ATAN_TBL
RL_COR_PEAK
IM_COR_PEAK
a
0.136
0.995
z^{-1}
SFCS_IC
(FROM INIT_AFC)
SFCS
2.5
-2.5
b
2.5
HFCS
(0-5V TO VCXO)

FIG. 35

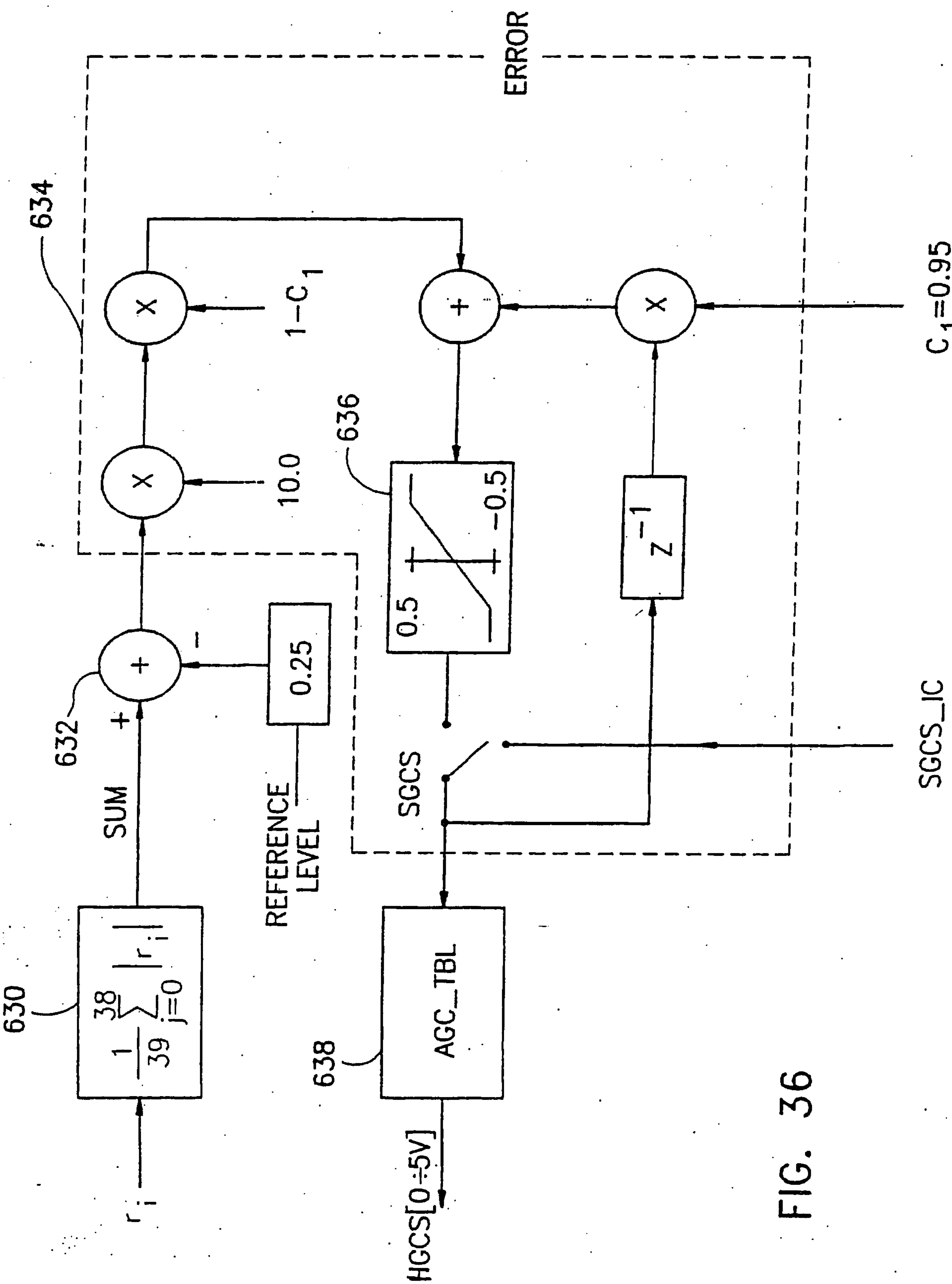

ERROR
634
1−C1
C1=0.95
10.0
636
0.5
−0.5
z−1
632
+
−
0.25
SUM
REFERENCE LEVEL
630
$\frac{1}{39}\sum_{j=0}^{38}|r_j|$
r_i
SGCS
SGCS_IC
AGC_TBL
638
HGCS[0÷5V]

FIG. 36

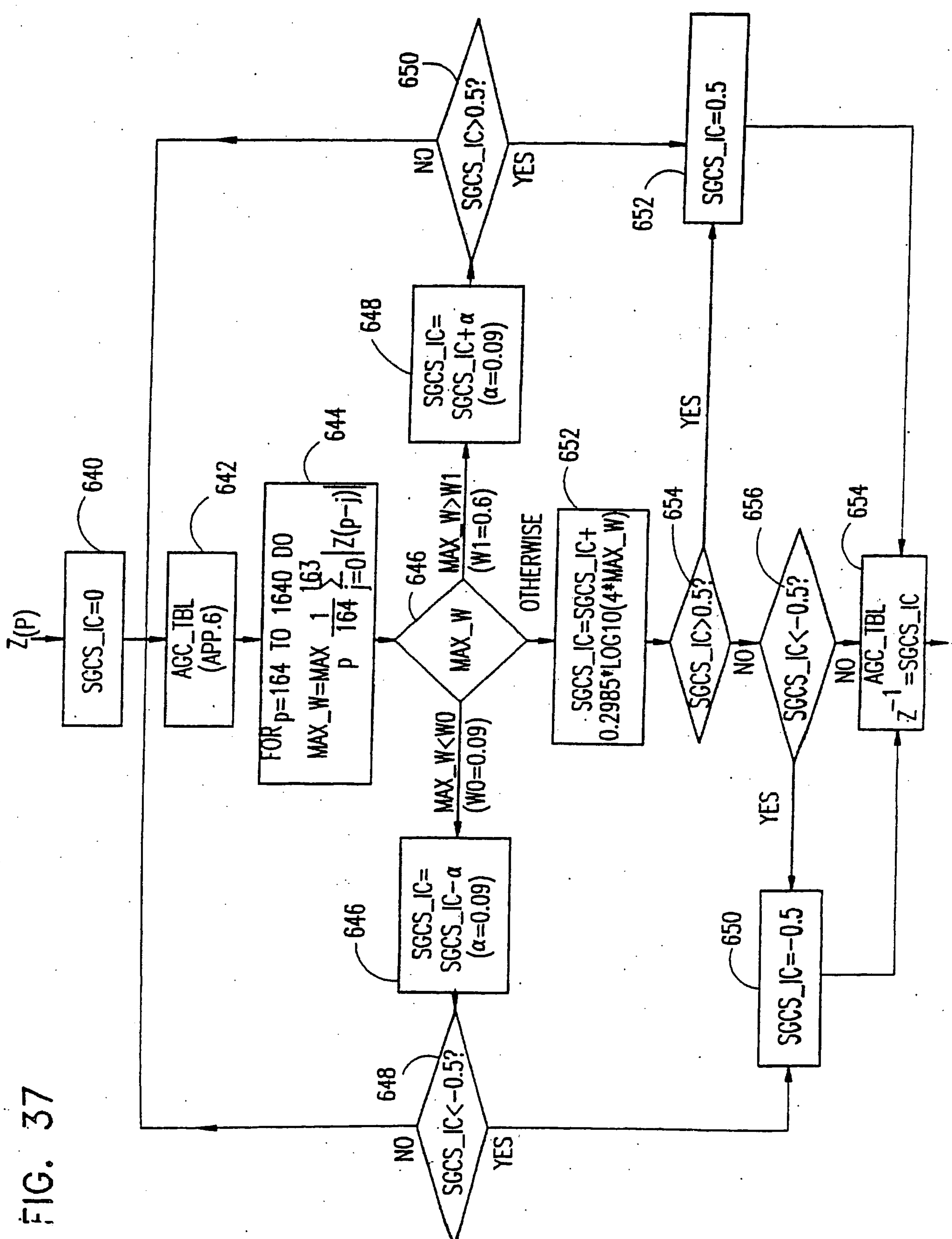

Z(P)
SGCS_IC=0
640
AGC_TBL (APP.6)
642
FOR p=164 TO 1640 DO MAX_W=MAX 1/164 Σ163 j=0 |Z(p−j)|
644
MAX_W
646
MAX_W>W1 (W1=0.6)
MAX_W<W0 (W0=0.09)
OTHERWISE
SGCS_IC= SGCS_IC+α (α=0.09)
648
SGCS_IC= SGCS_IC−α (α=0.09)
646
SGCS_IC>0.5?
650
SGCS_IC<−0.5?
648
NO
YES
SGCS_IC=0.5
652
SGCS_IC=−0.5
650
SGCS_IC=SGCS_IC+ 0.2985*LOG10(4*MAX_W)
652
SGCS_IC>0.5?
654
SGCS_IC<−0.5?
656
AGC_TBL Z−1 =SGCS_IC
654

FIG. 37

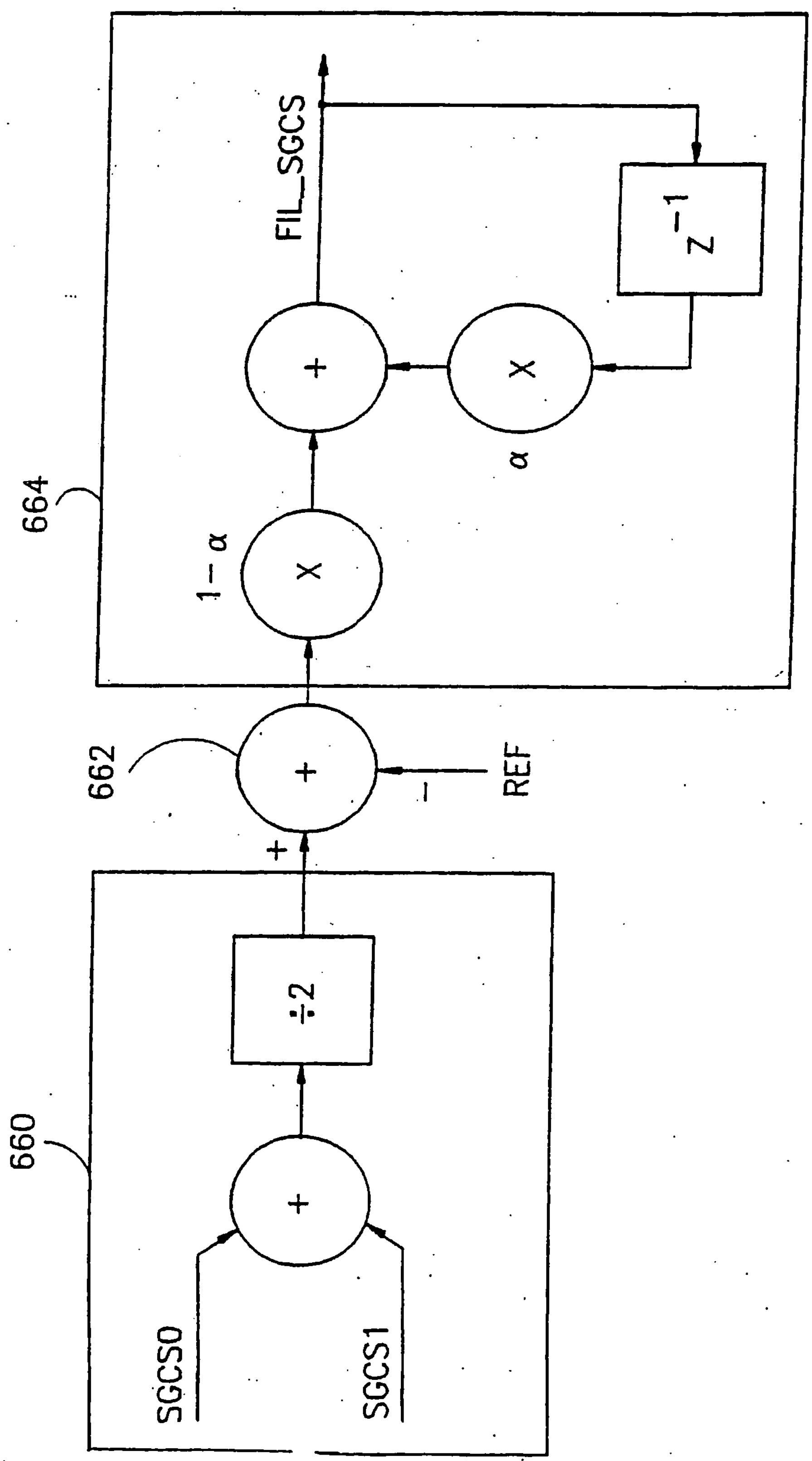


FIG. 38

FIG. 39

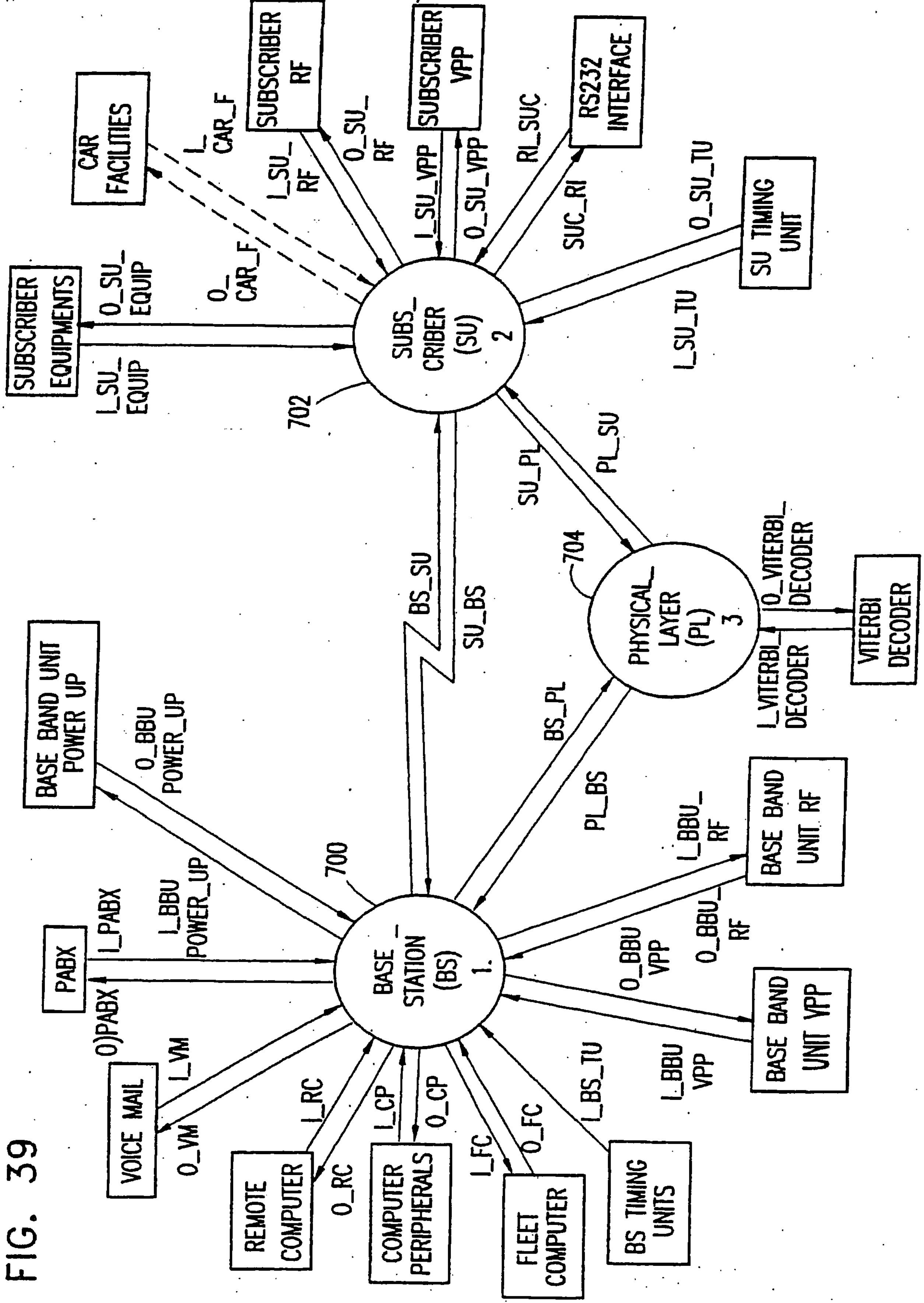
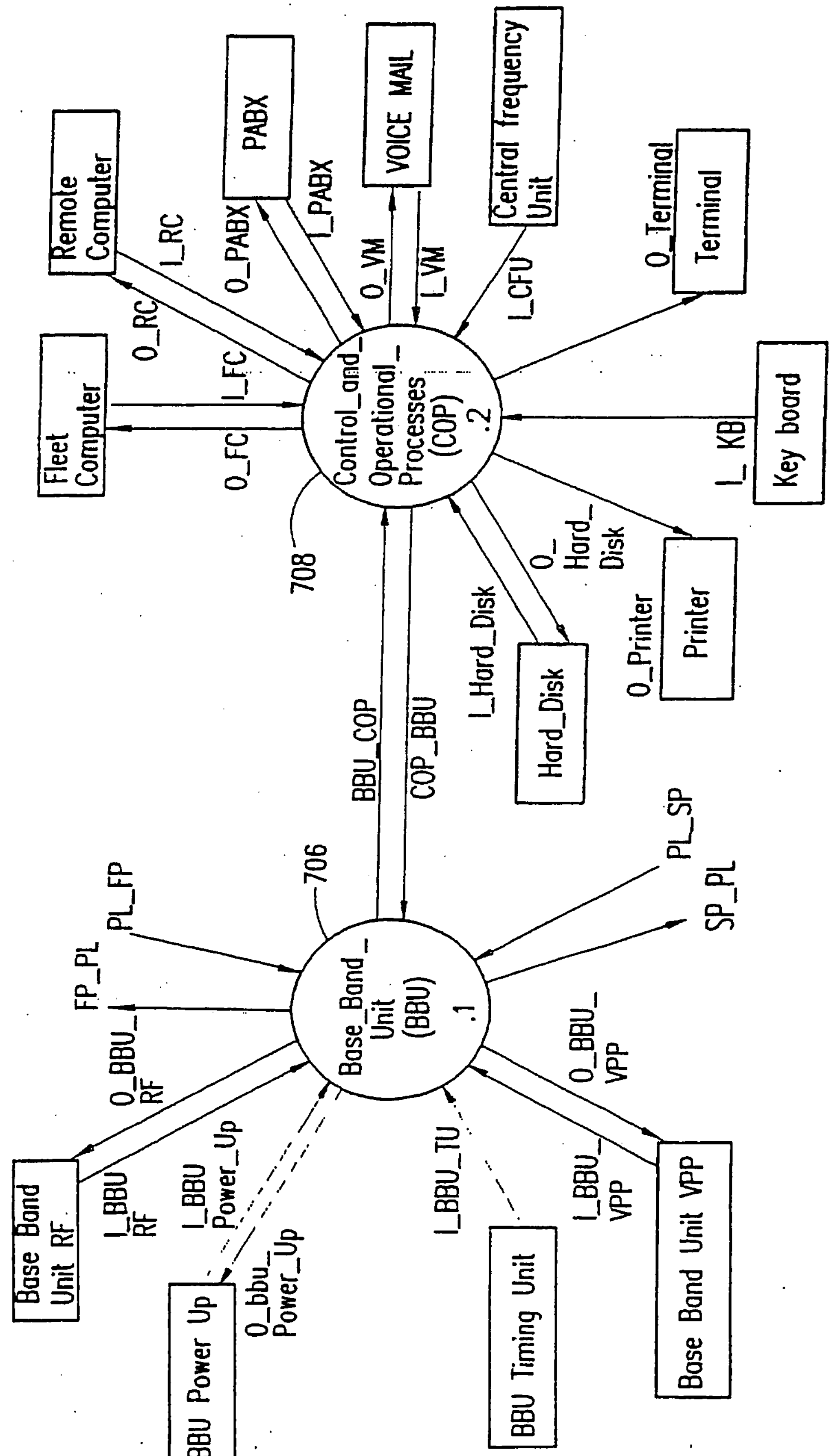

CAR FACILITIES
SUBSCRIBER RF
SUBSCRIBER VPP
RS232 INTERFACE
SU TIMING UNIT
SUBSCRIBER EQUIPMENTS
SUBS_ CRIBER (SU) 2
PHYSICAL_ LAYER (PL) 3
VITERBI DECODER
BASE BAND UNIT POWER UP
BASE BAND UNIT RF
BASE BAND UNIT VPP
BASE_ STATION (BS) 1.
PABX
VOICE MAIL
REMOTE COMPUTER
COMPUTER PERIPHERALS
FLEET COMPUTER
BS TIMING UNITS
I_CAR_F
O_CAR_F
I_SU_RF
O_SU_RF
I_SU_VPP
O_SU_VPP
RI_SUC
SUC_RI
O_SU_TU
I_SU_TU
O_SU_EQUIP
I_SU_EQUIP
702
SU_PL
PL_SU
704
O_VITERBI_DECODER
I_VITERBI_DECODER
BS_SU
SU_BS
BS_PL
PL_BS
O_BBU POWER_UP
I_BBU POWER_UP
700
I_BBU_RF
O_BBU_RF
O_BBU VPP
I_BBU VPP
I_PABX
O)PABX
I_VM
O_VM
I_RC
O_RC
I_CP
O_CP
I_FC
O_FC
I_BS_TU

FIG. 40



Remote Computer
PABX
VOICE MAIL
Central frequency Unit
Terminal
Key board
Printer
Hard_Disk
Fleet Computer
I_RC
O_RC
O_PABX
I_PABX
O_VM
I_VM
I_CFU
O_Terminal
I_FC
O_FC
I_KB
O_Hard_Disk
I_Hard_Disk
O_Printer
Control_and_Operational_Processes (COP) .2
708
BBU_COP
COP_BBU
706
Base_Band_Unit (BBU) .1
FP_PL
PL_FP
PL_SP
SP_PL
O_BBU_RF
I_BBU_RF
I_BBU Power_Up
O_bbu_Power_Up
I_BBU_TU
O_BBU_VPP
I_BBU_VPP
Base Band Unit RF
BBU Power Up
BBU Timing Unit
Base Band Unit VPP

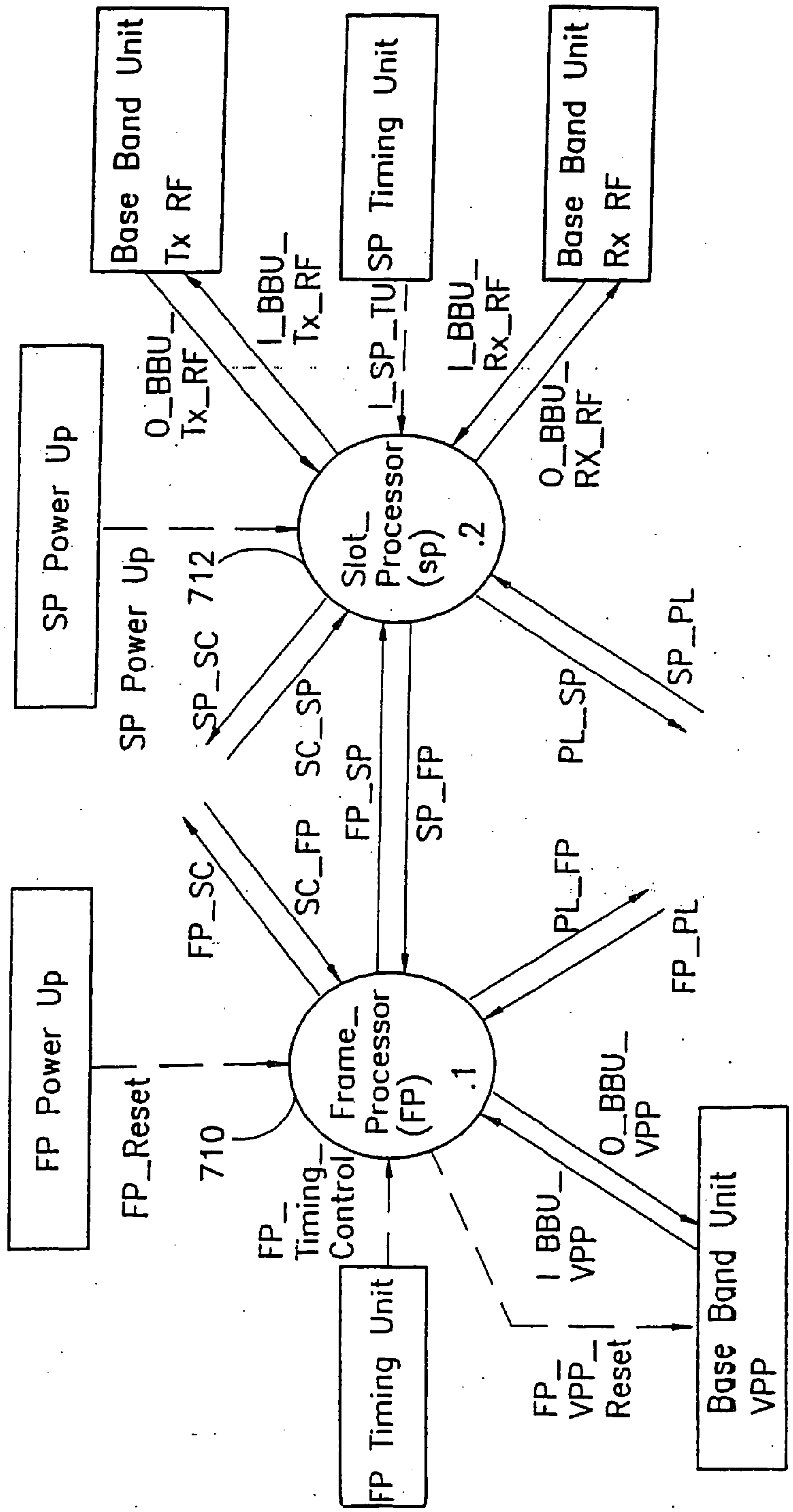

Base Band Unit
Tx RF
SP Timing Unit
Base Band Unit
Rx RF
SP Power Up
O_BBU_
Tx_RF
I_BBU_
Tx_RF
I_SP_TU
I_BBU_
Rx_RF
O_BBU_
RX_RF
Slot_
Processor
(sp)
.2
SP Power Up
712
SP_SC
SC_SP
SP_PL
PL_SP
FP_SP
SP_FP
FP_SC
SC_FP
PL_FP
FP_PL
FP Power Up
FP_Reset
710
Frame_
Processor
(FP)
.1
FP_
Timing_
Control
FP Timing Unit
I BBU_
VPP
O_BBU_
VPP
FP_
VPP_
Reset
Base Band Unit
VPP

FIG. 41

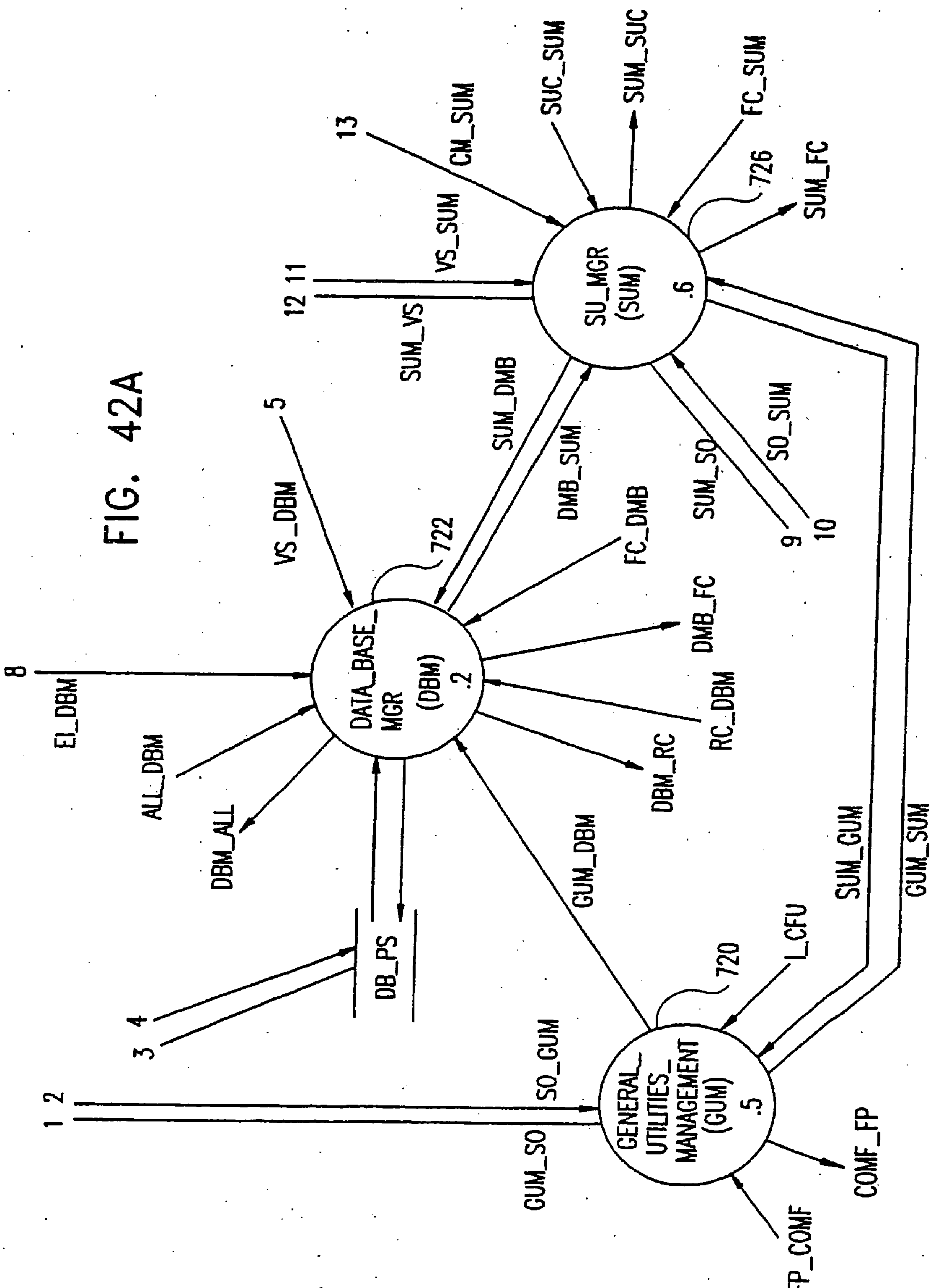

SU_MGR
(SUM)
.6
DATA_BASE_
MGR
(DBM)
.2
GENERAL_
UTILITIES_
MANAGEMENT
(GUM)
.5
DB_PS
13
CM_SUM
SUC_SUM
SUM_SUC
FC_SUM
726
SUM_FC
12 11
VS_SUM
SUM_VS
SUM_DMB
DMB_SUM
SO_SUM
SUM_SO
9
10
5
VS_DBM
722
FC_DMB
DMB_FC
RC_DBM
DBM_RC
8
EL_DBM
ALL_DBM
DBM_ALL
GUM_DBM
SUM_GUM
GUM_SUM
I_CFU
720
3 4
1 2
SO_GUM
GUM_SO
COMF_FP
FP_COMF

FIG. 42A

FIG. 42B

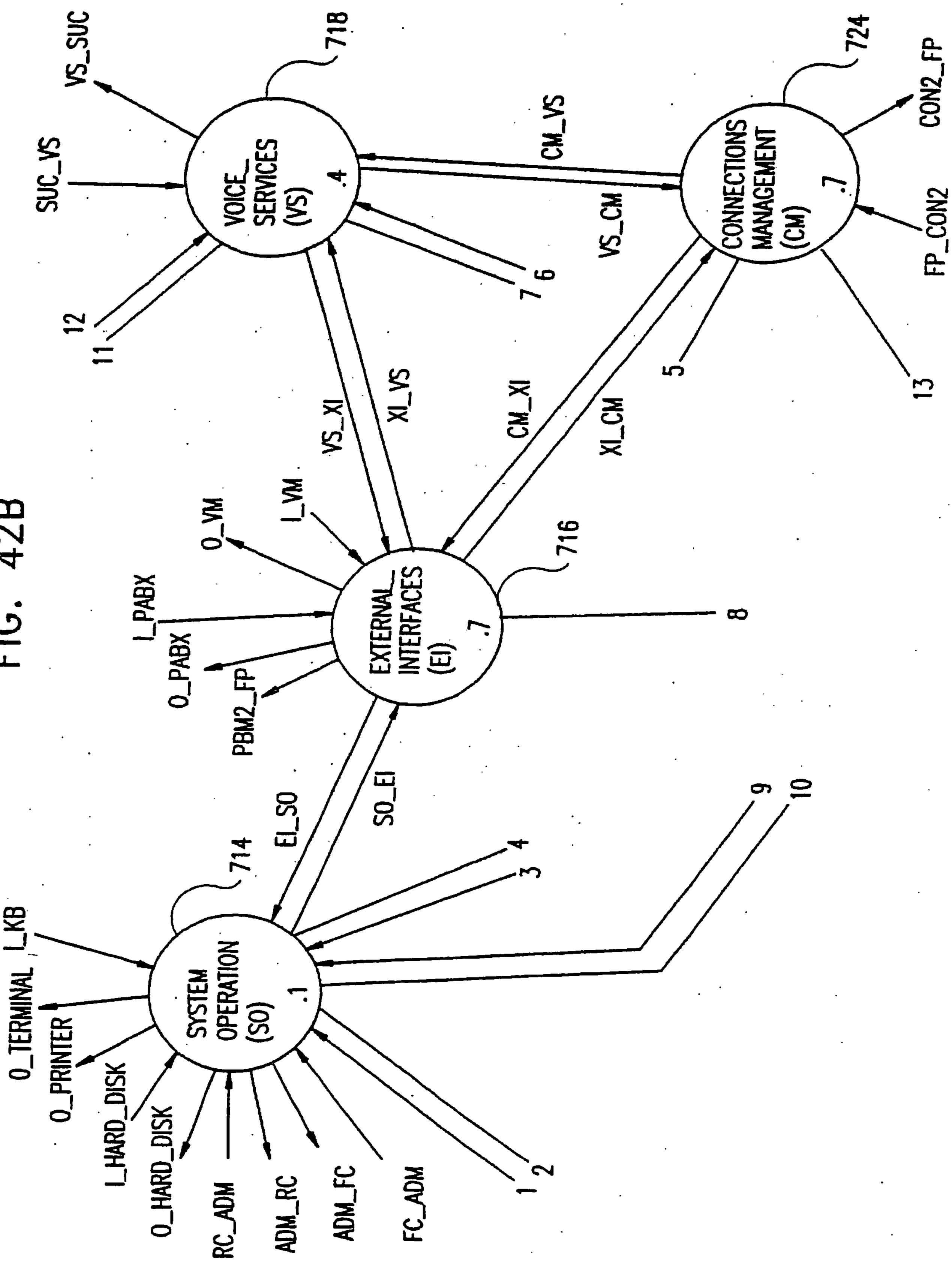
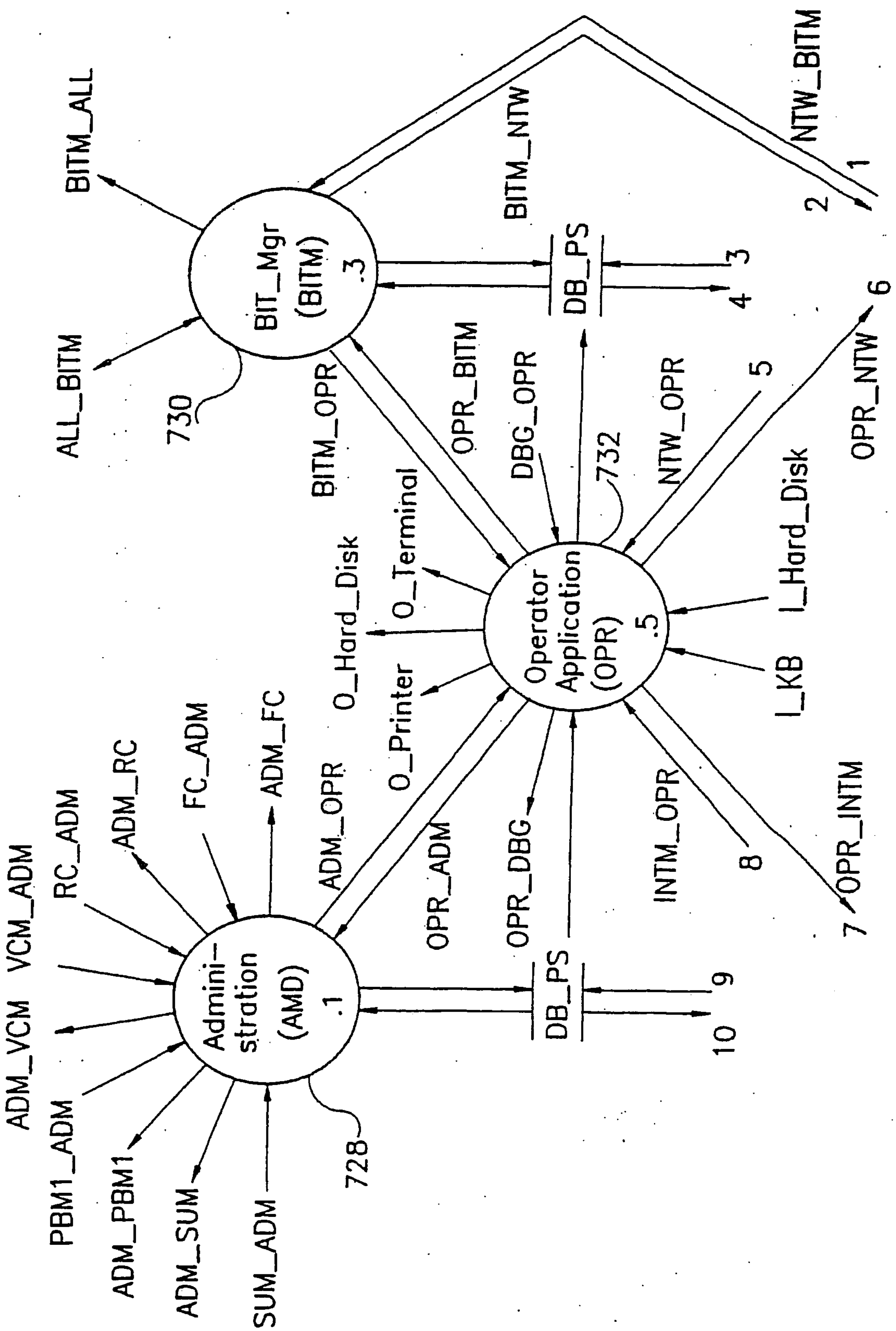

VOICE_ SERVICES (VS) .4
718
VS_SUC
SUC_VS
12
11
7
6
CM_VS
VS_CM
CONNECTIONS MANAGEMENT (CM) .7
724
CON2_FP
FP_CON2
5
13
VS_XI
XI_VS
CM_XI
XI_CM
EXTERNAL_ INTERFACES (EI) .7
716
O_VM
I_VM
I_PABX
O_PABX
PBM2_FP
8
EI_SO
SO_EI
SYSTEM OPERATION (SO) .1
714
4
3
9
10
I_KB
O_TERMINAL
O_PRINTER
I_HARD_DISK
O_HARD_DISK
RC_ADM
ADM_RC
ADM_FC
FC_ADM
1
2

FIG.43A


BITM_ALL
ALL_BITM
BIT_Mgr
(BITM)
.3
730
BITM_NTW
NTW_BITM
1
2
DB_PS
3
4
6
5
BITM_OPR
OPR_BITM
DBG_OPR
732
NTW_OPR
OPR_NTW
I_Hard_Disk
O_Terminal
O_Hard_Disk
O_Printer
Operator
Application
(OPR)
.5
I_KB
INTM_OPR
8
OPR_INTM
7
OPR_DBG
OPR_ADM
ADM_OPR
ADM_FC
FC_ADM
ADM_RC
RC_ADM
VCM_ADM
ADM_VCM
PBM1_ADM
ADM_PBM1
ADM_SUM
SUM_ADM
Admini-
stration
(AMD)
.1
728
DB_PS
9
10

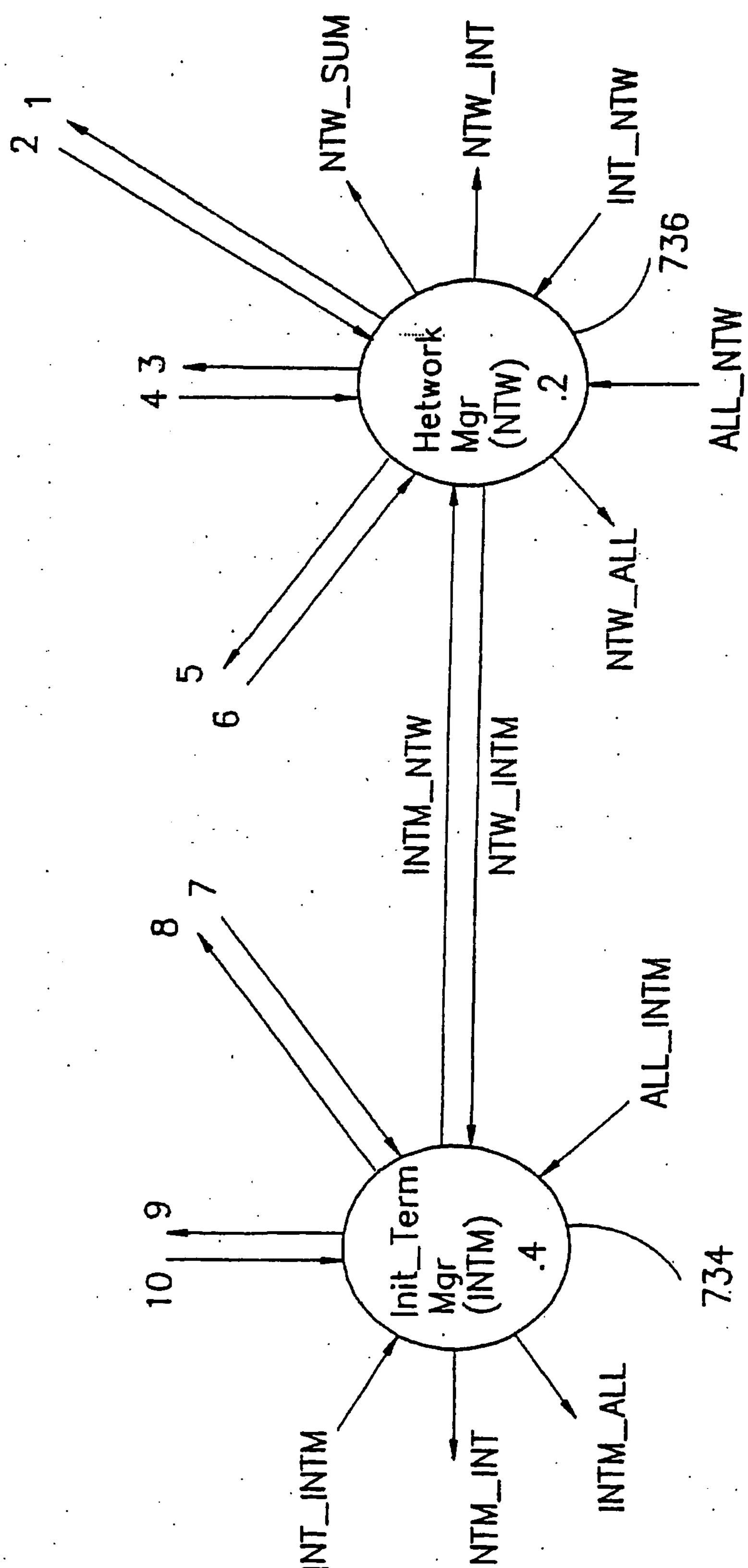

1
2
NTW_SUM
NTW_INT
INT_NTW
736
3
4
Hetwork
Mgr
(NTW)
.2
ALL_NTW
5
6
NTW_ALL
INTM_NTW
NTW_INTM
7
8
ALL_INTM
9
10
Init_Term
Mgr
(INTM)
.4
734
INT_INTM
INTM_INT
INTM_ALL

FIG. 43B

FIG. 44

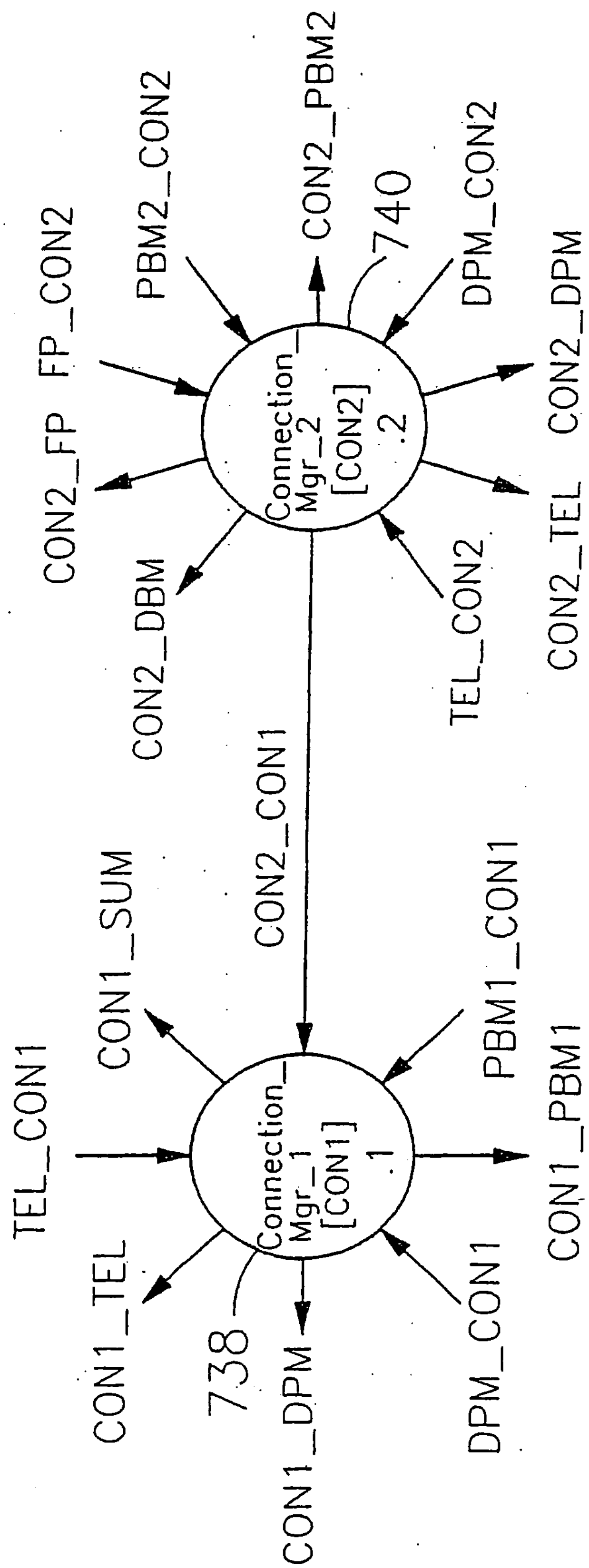
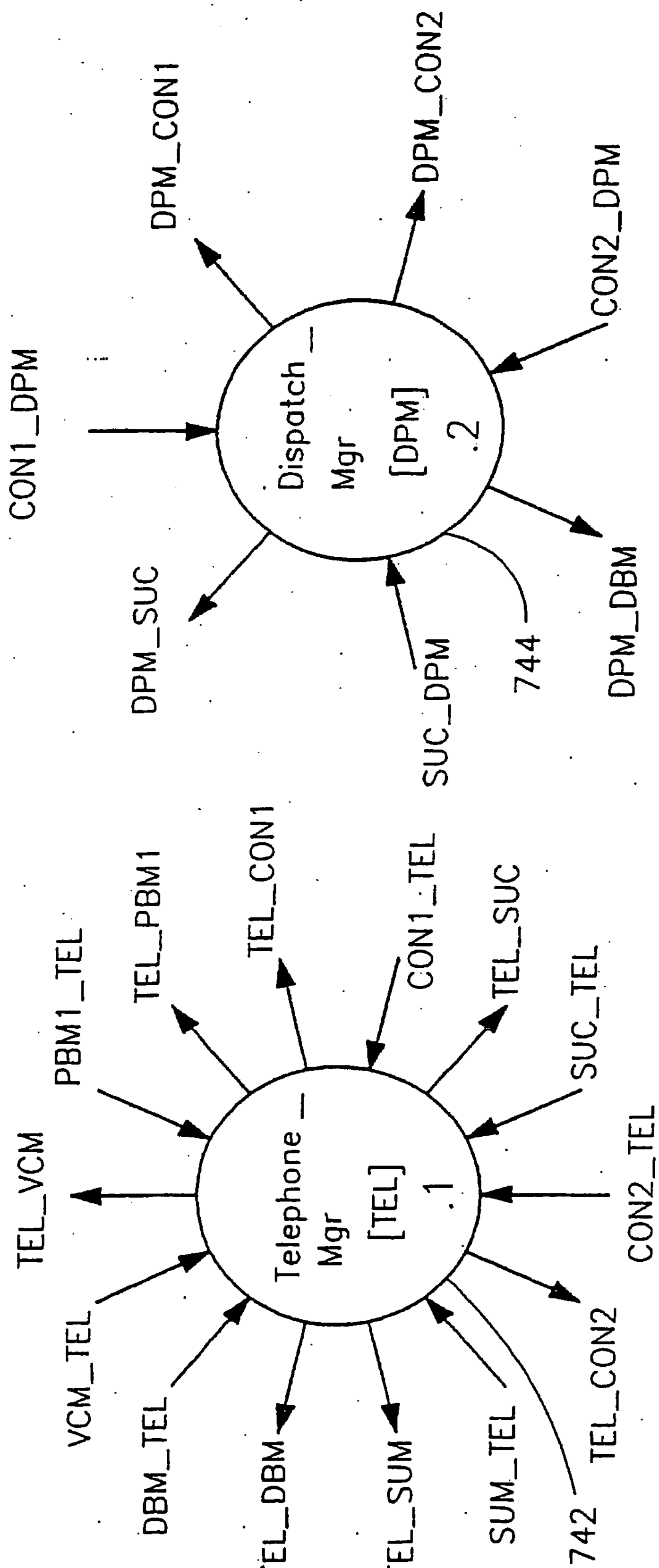

Connection_ Mgr_2 [CON2] .2
Connection_ Mgr_1 [CON1] .1
740
738
FP_CON2
CON2_FP
PBM2_CON2
CON2_PBM2
DPM_CON2
CON2_DPM
CON2_TEL
TEL_CON2
CON2_DBM
CON2_CON1
TEL_CON1
CON1_SUM
PBM1_CON1
CON1_PBM1
DPM_CON1
CON1_DPM
CON1_TEL

FIG. 45


Dispatch Mgr [DPM] .2
CON1_DPM
DPM_CON1
DPM_CON2
CON2_DPM
DPM_DBM
744
SUC_DPM
DPM_SUC
Telephone Mgr [TEL] .1
TEL_VCM
PBM1_TEL
TEL_PBM1
TEL_CON1
CON1_TEL
TEL_SUC
SUC_TEL
CON2_TEL
TEL_CON2
742
SUM_TEL
TEL_SUM
TEL_DBM
DBM_TEL
VCM_TEL

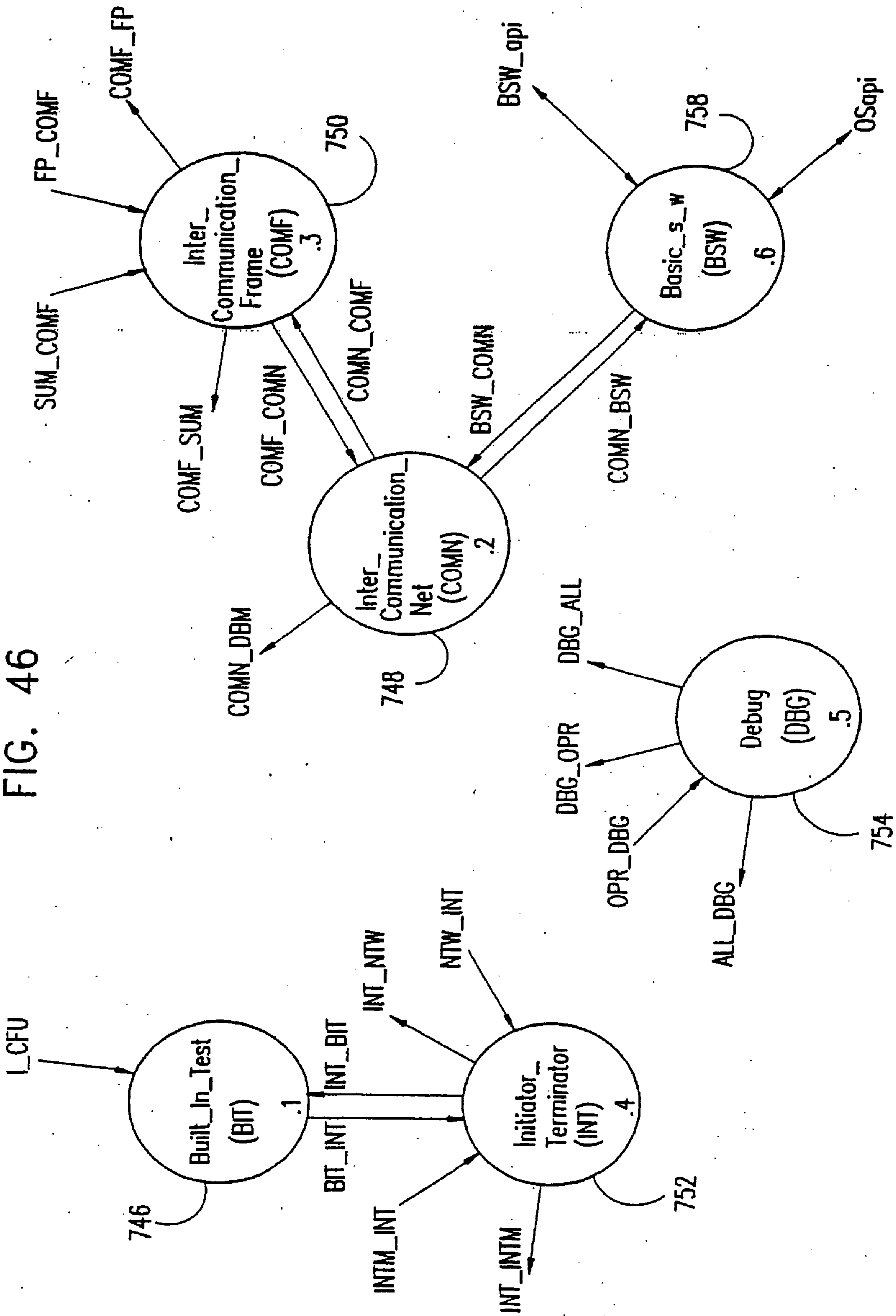

Inter_
Communication_
Frame
(COMF)
.3
750
FP_COMF
COMF_FP
SUM_COMF
COMF_SUM
COMF_COMN
COMN_COMF
Basic_s_w
(BSW)
.6
758
BSW_api
OSapi
BSW_COMN
COMN_BSW
Inter_
Communication_
Net
(COMN)
.2
748
COMN_DBM
Debug
(DBG)
.5
754
DBG_ALL
DBG_OPR
OPR_DBG
ALL_DBG
Built_In_Test
(BIT)
.1
746
I_CFU
INT_BIT
BIT_INT
Initiator_
Terminator
(INT)
.4
752
INT_NTW
NTW_INT
INTM_INT
INT_INTM

FIG. 46

FIG. 47

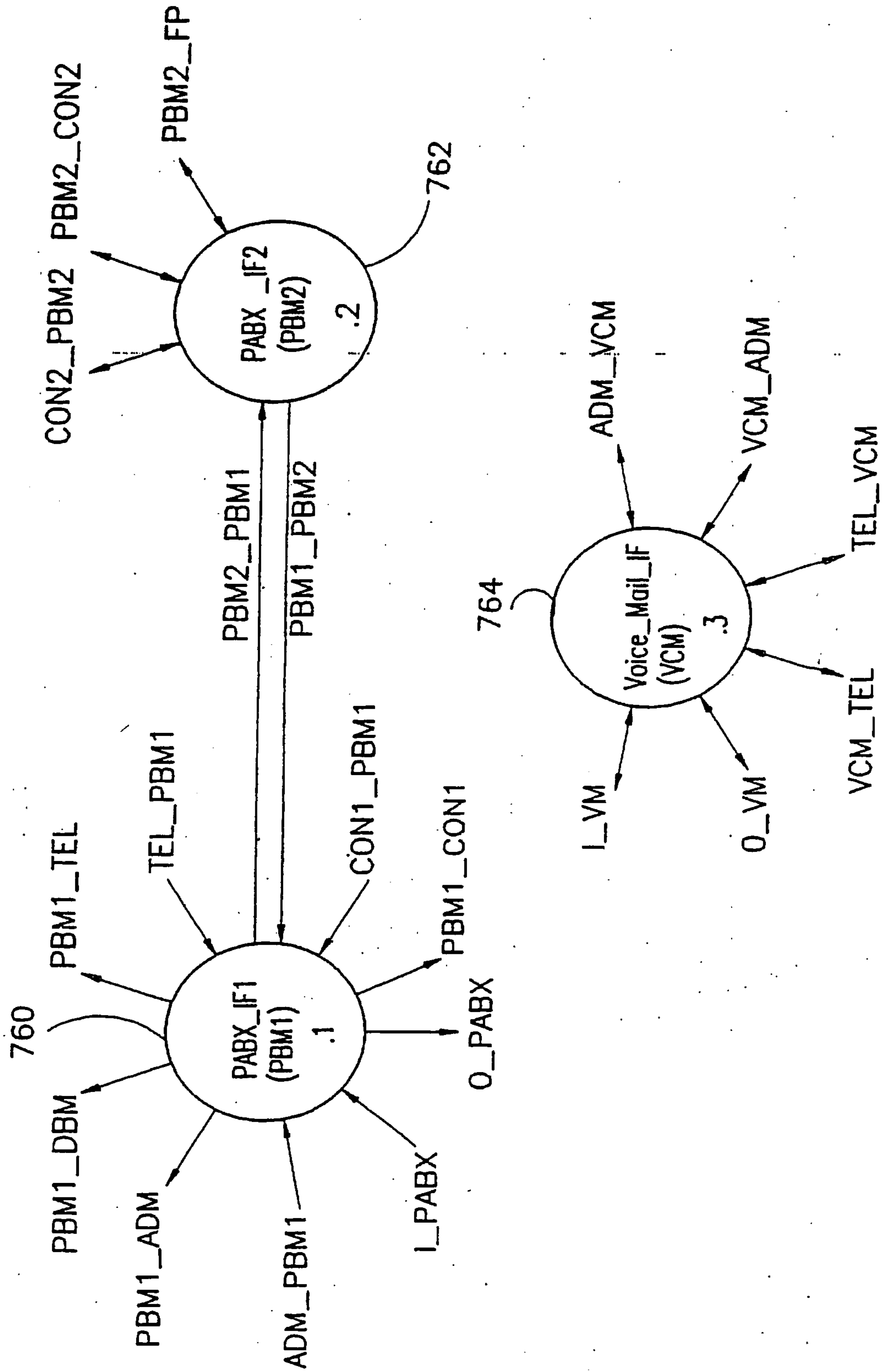
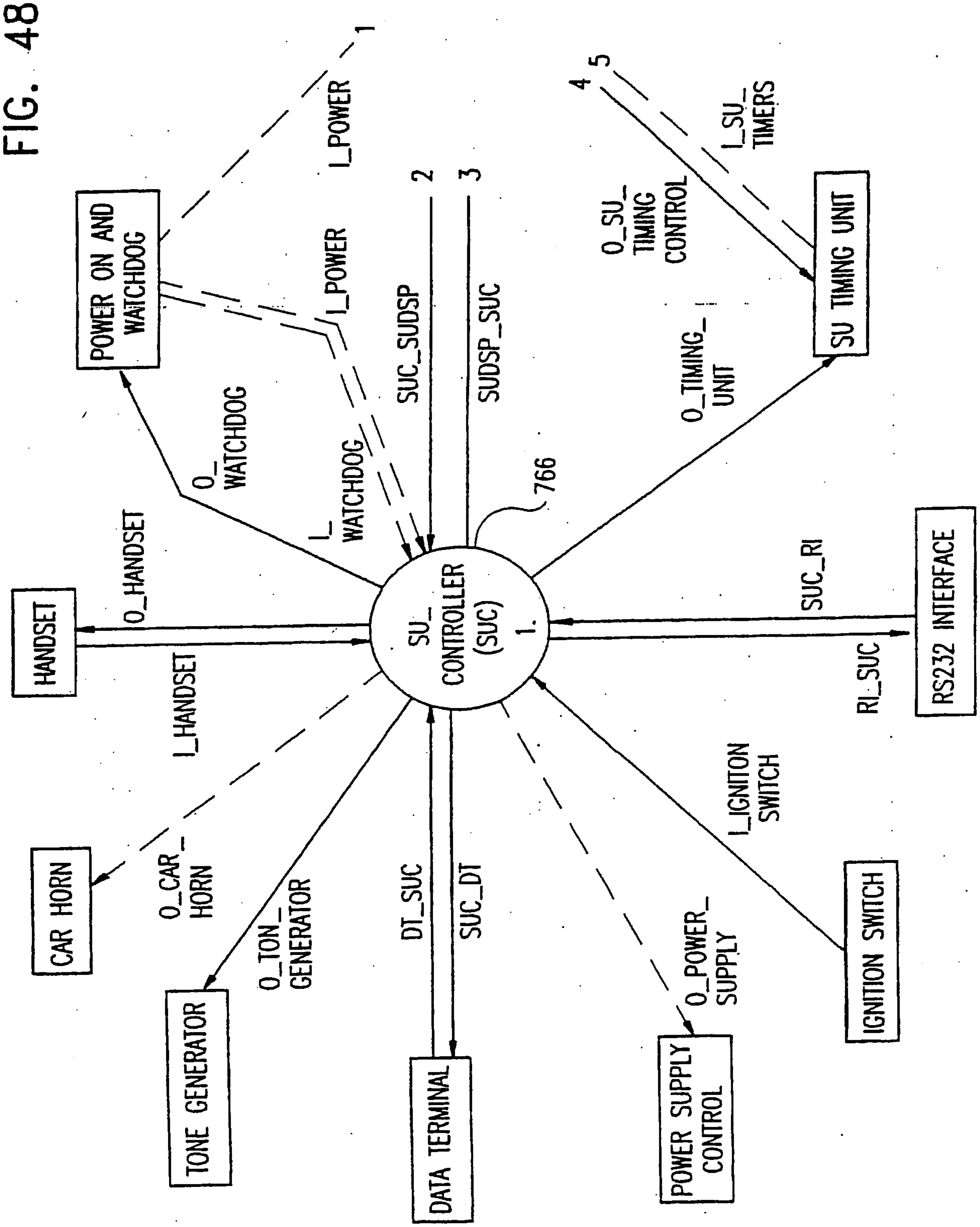

PABX_IF1
(PBM1)
.1
760
PBM1_TEL
TEL_PBM1
CON1_PBM1
PBM1_CON1
O_PABX
I_PABX
ADM_PBM1
PBM1_ADM
PBM1_DBM
PBM2_PBM1
PBM1_PBM2
PABX _IF2
(PBM2)
.2
762
CON2_PBM2
PBM2_CON2
PBM2_FP
Voice_Mail_IF
(VCM)
.3
764
ADM_VCM
VCM_ADM
TEL_VCM
VCM_TEL
O_VM
I_VM

FIG. 48A


POWER ON AND WATCHDOG
I_POWER
1
I_POWER
O_WATCHDOG
I_WATCHDOG
SUC_SUDSP
2
SUDSP_SUC
3
4
5
O_SU_TIMING CONTROL
I_SU_TIMERS
SU TIMING UNIT
O_TIMING_UNIT
766
SU_CONTROLLER (SUC) 1.
HANDSET
O_HANDSET
I_HANDSET
SUC_RI
RI_SUC
RS232 INTERFACE
CAR HORN
O_CAR_HORN
TONE GENERATOR
O_TON_GENERATOR
DT_SUC
SUC_DT
DATA TERMINAL
O_POWER_SUPPLY
POWER SUPPLY CONTROL
I_IGNITON SWITCH
IGNITION SWITCH

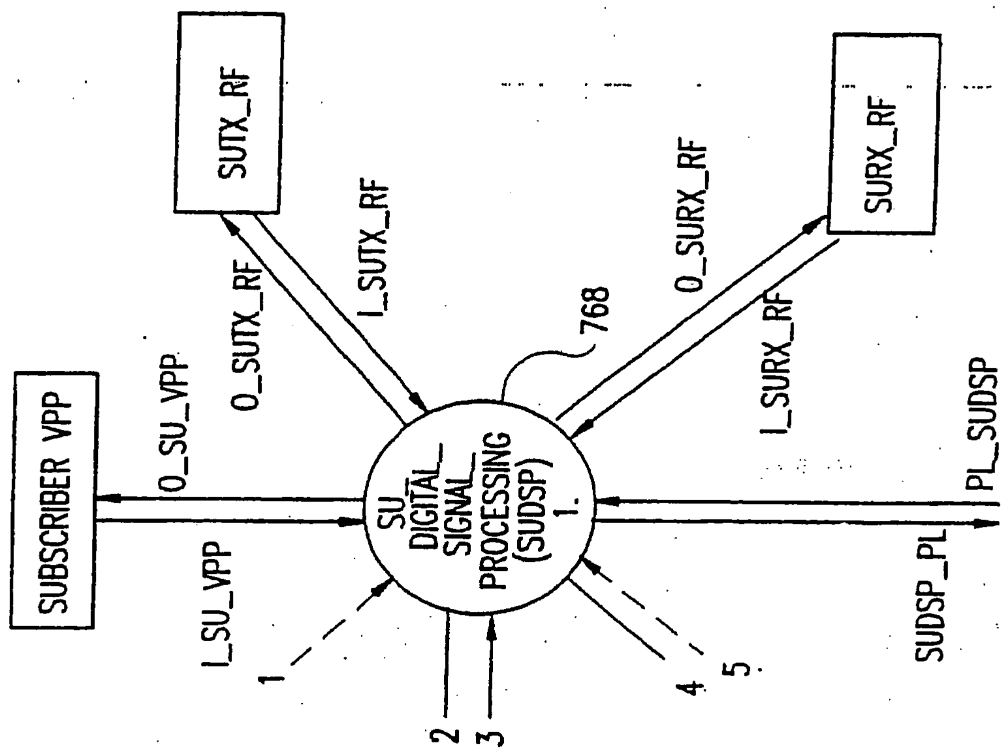

SUBSCRIBER VPP
O_SU_VPP
I_SU_VPP
SUTX_RF
O_SUTX_RF
I_SUTX_RF
SURX_RF
O_SURX_RF
I_SURX_RF
768
SU_
DIGITAL_
SIGNAL_
PROCESSING
(SUDSP)
1.
PL_SUDSP
SUDSP_PL
1
2
3
4
5

FIG. 48B

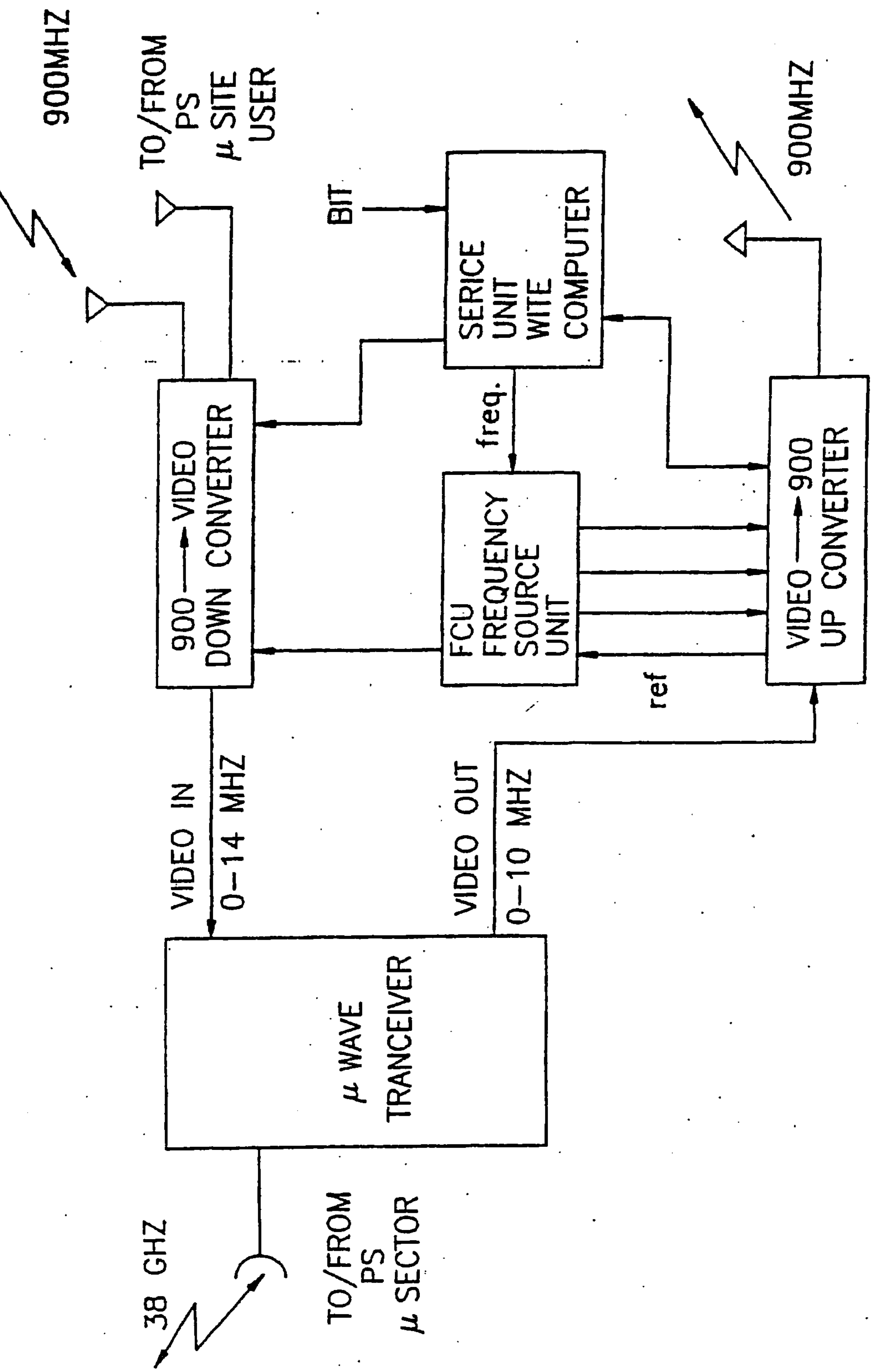

900MHZ
TO/FROM
PS
μ SITE
USER
BIT
SERICE
UNIT
WITE
COMPUTER
900MHZ
freq.
900 ⟶ VIDEO
DOWN CONVERTER
FCU
FREQUENCY
SOURCE
UNIT
VIDEO ⟶ 900
UP CONVERTER
ref
VIDEO IN
0–14 MHZ
VIDEO OUT
0–10 MHZ
μ WAVE
TRANCEIVER
38 GHZ
TO/FROM
PS
μ SECTOR

FIG. 49

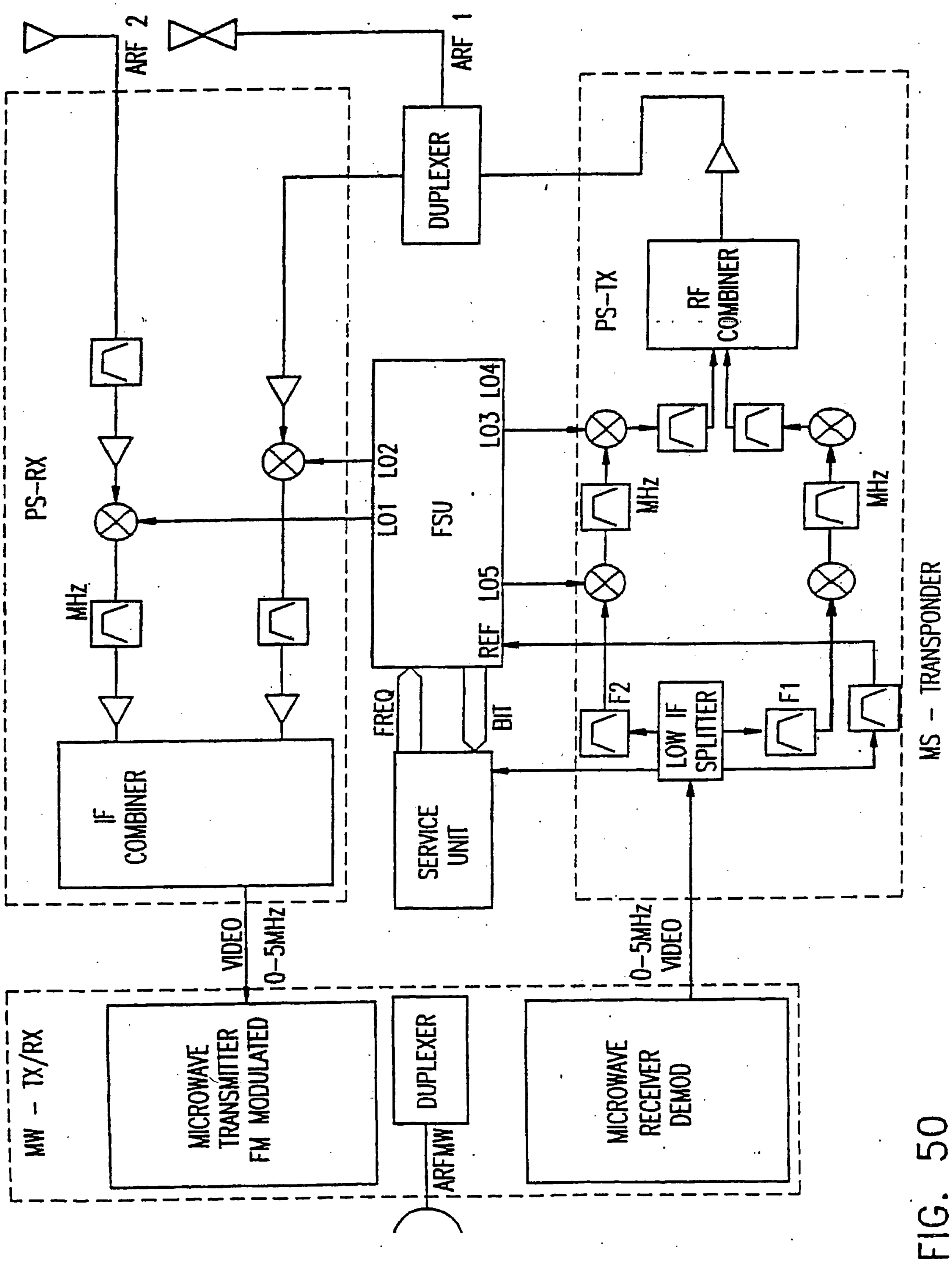

ARF 2
ARF 1
DUPLEXER
PS-RX
PS-TX
RF COMBINER
FSU
LO1
LO2
LO3
LO4
LO5
REF
MHz
MHz
MHz
IF COMBINER
FREQ
BIT
SERVICE UNIT
LOW IF SPLITTER
F2
F1
MS - TRANSPONDER
VIDEO
0-5MHz
0-5MHz
VIDEO
MW - TX/RX
MICROWAVE TRANSMITTER FM MODULATED
DUPLEXER
ARFMW
MICROWAVE RECEIVER DEMOD

FIG. 50

FIG. 51

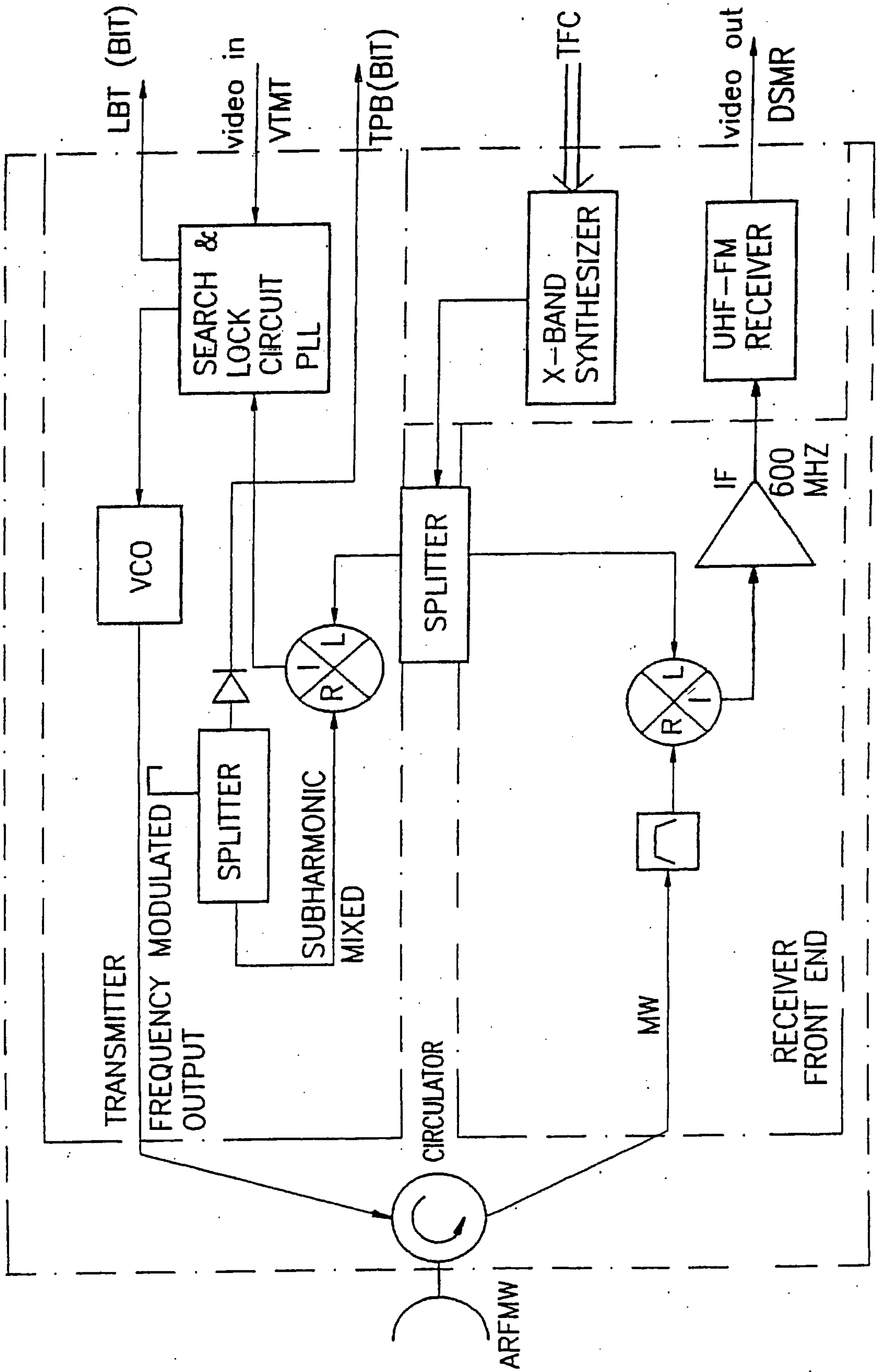
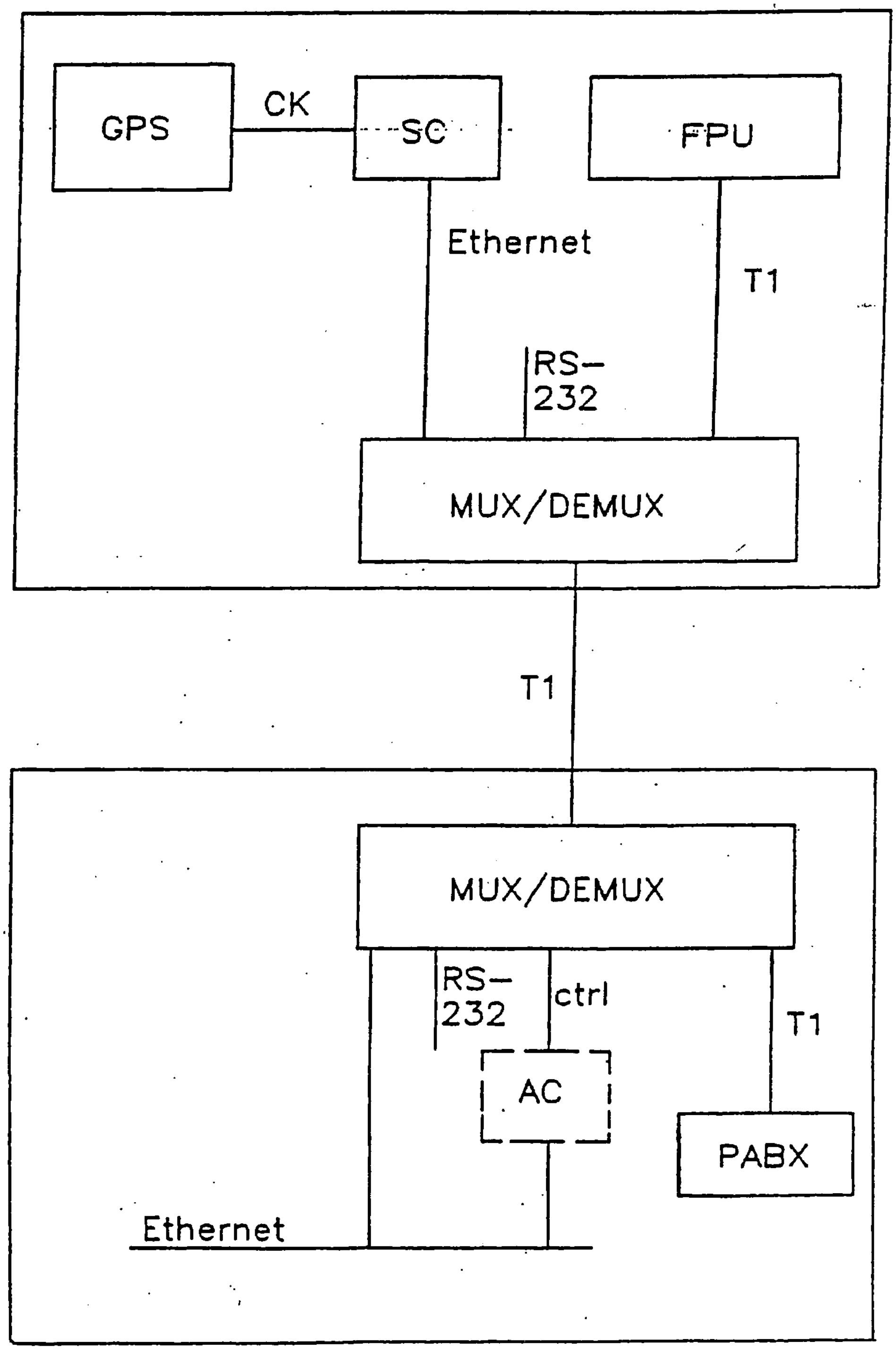

LBT (BIT)
video in
VTMT
TPB(BIT)
TFC
video out
DSMR
SEARCH & LOCK CIRCUIT PLL
X-BAND SYNTHESIZER
UHF-FM RECEIVER
VCO
SPLITTER
IF
600 MHZ
SPLITTER
SUBHARMONIC MIXED
TRANSMITTER
FREQUENCY MODULATED OUTPUT
CIRCULATOR
MW
RECEIVER FRONT END
ARFMW

FIG. 52

Remote Sector

Base Station



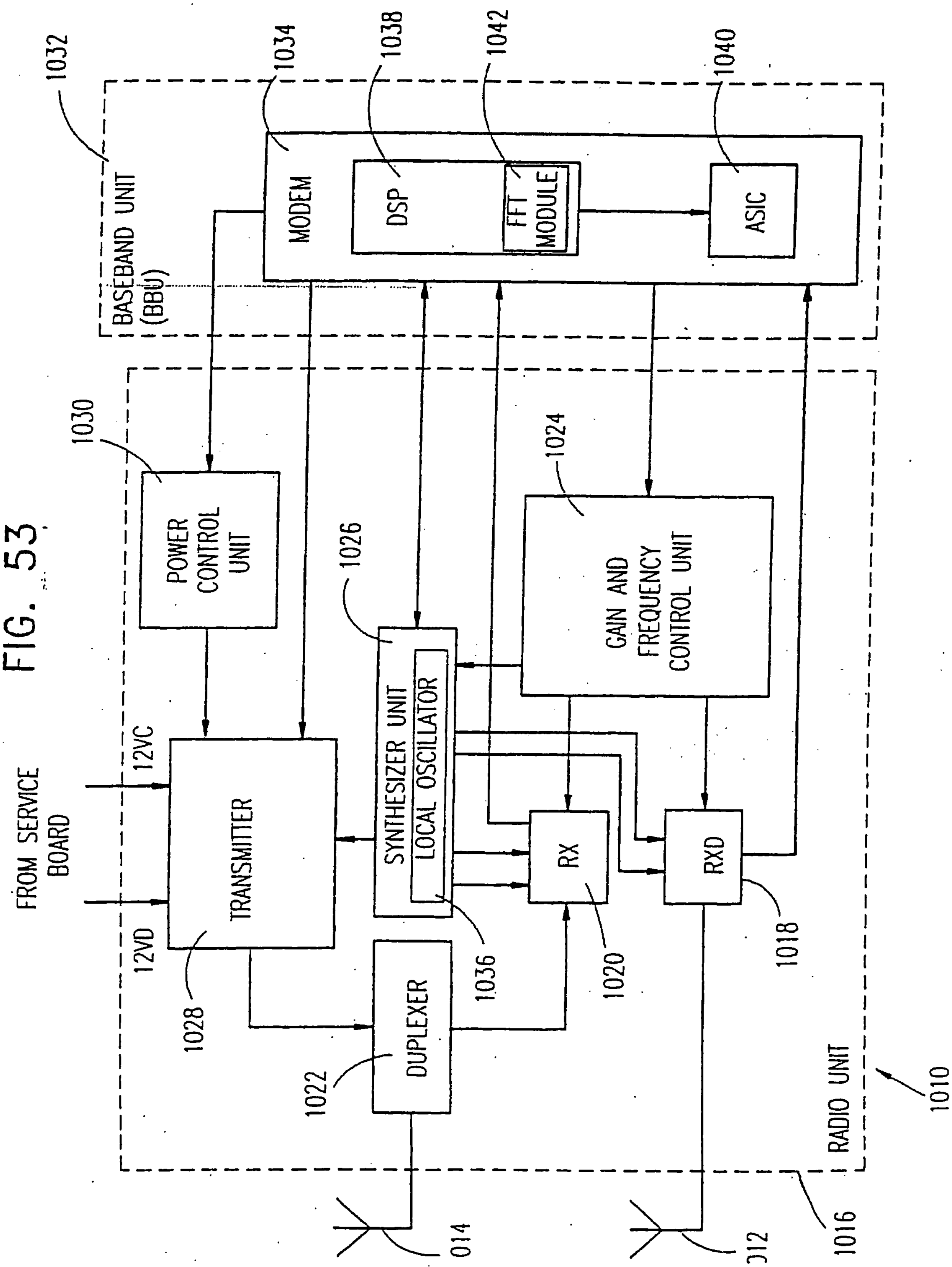
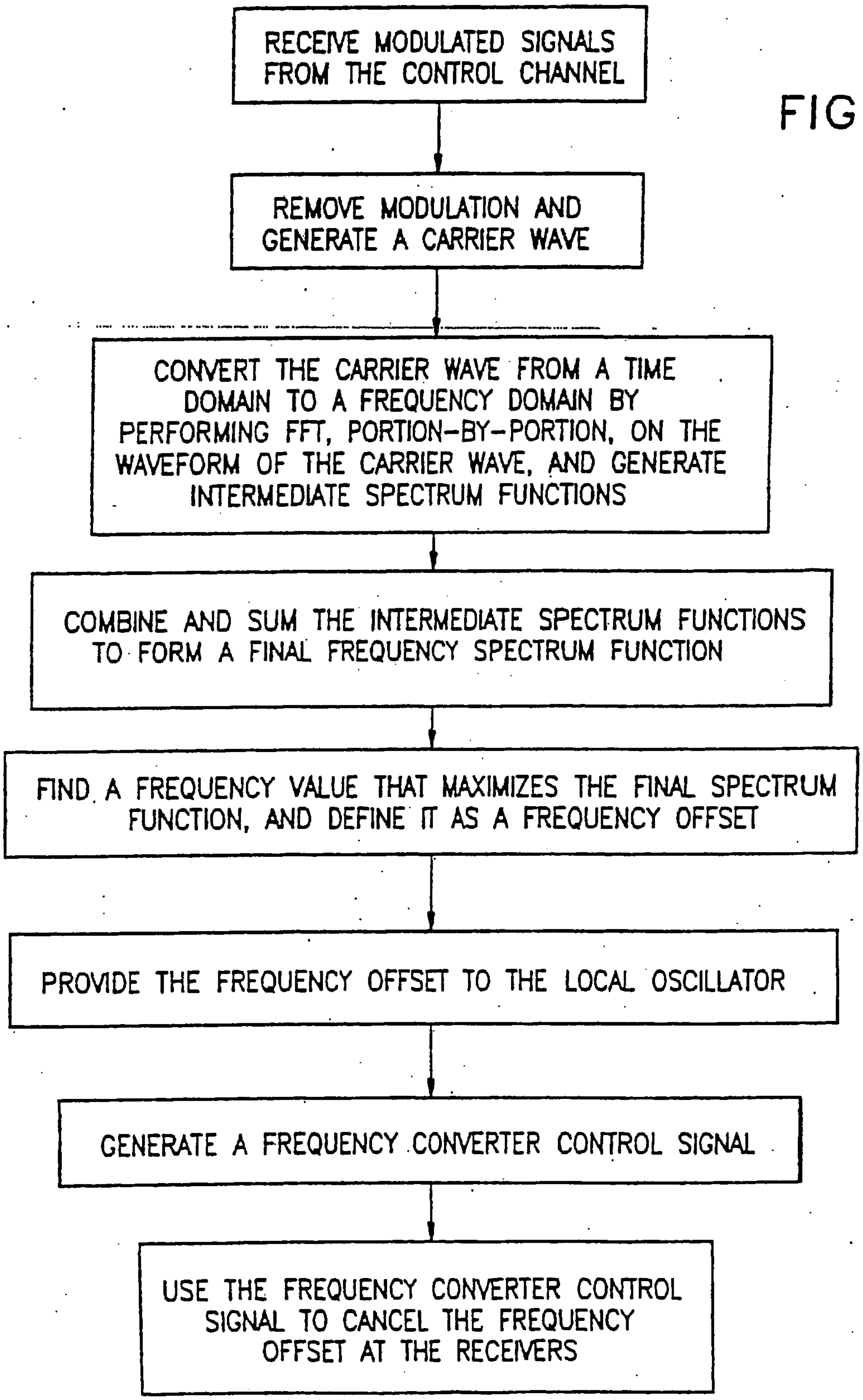

BASEBAND UNIT (BBU)
1032
MODEM
1034
DSP
1038
FFT MODULE
1042
ASIC
1040
POWER CONTROL UNIT
1030
GAIN AND FREQUENCY CONTROL UNIT
1024
SYNTHESIZER UNIT
LOCAL OSCILLATOR
1026
1036
FROM SERVICE BOARD
12VC
12VD
TRANSMITTER
1028
DUPLEXER
1022
RX
1020
RXD
1018
RADIO UNIT
1010
1016
014
012

FIG. 53

FIG. 54


RECEIVE MODULATED SIGNALS FROM THE CONTROL CHANNEL
REMOVE MODULATION AND GENERATE A CARRIER WAVE
CONVERT THE CARRIER WAVE FROM A TIME DOMAIN TO A FREQUENCY DOMAIN BY PERFORMING FFT, PORTION-BY-PORTION, ON THE WAVEFORM OF THE CARRIER WAVE, AND GENERATE INTERMEDIATE SPECTRUM FUNCTIONS
COMBINE AND SUM THE INTERMEDIATE SPECTRUM FUNCTIONS TO FORM A FINAL FREQUENCY SPECTRUM FUNCTION
FIND A FREQUENCY VALUE THAT MAXIMIZES THE FINAL SPECTRUM FUNCTION, AND DEFINE IT AS A FREQUENCY OFFSET
PROVIDE THE FREQUENCY OFFSET TO THE LOCAL OSCILLATOR
GENERATE A FREQUENCY CONVERTER CONTROL SIGNAL
USE THE FREQUENCY CONVERTER CONTROL SIGNAL TO CANCEL THE FREQUENCY OFFSET AT THE RECEIVERS

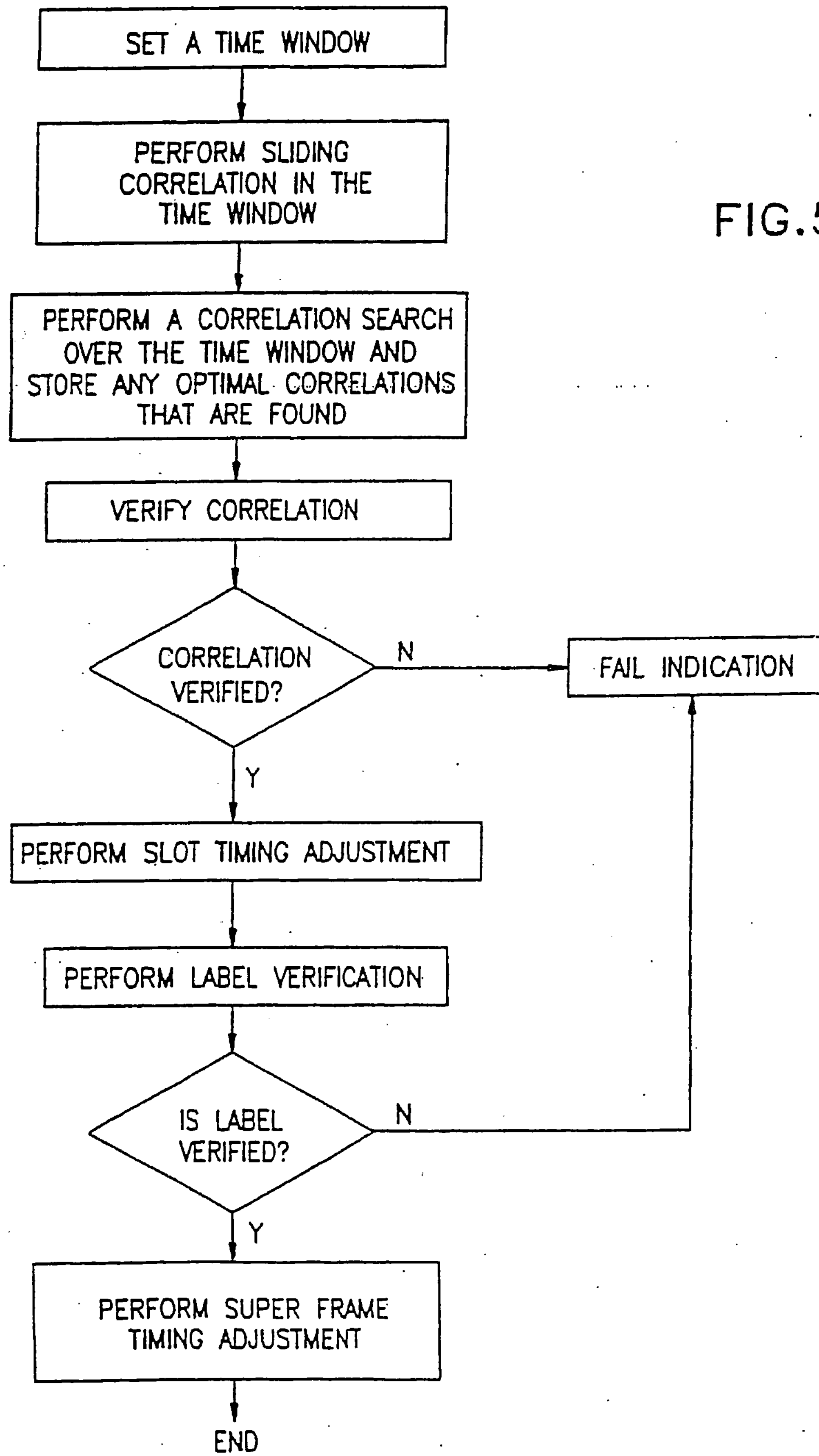

SET A TIME WINDOW
PERFORM SLIDING CORRELATION IN THE TIME WINDOW
PERFORM A CORRELATION SEARCH OVER THE TIME WINDOW AND STORE ANY OPTIMAL CORRELATIONS THAT ARE FOUND
VERIFY CORRELATION
CORRELATION VERIFIED?
N
FAIL INDICATION
Y
PERFORM SLOT TIMING ADJUSTMENT
PERFORM LABEL VERIFICATION
IS LABEL VERIFIED?
N
Y
PERFORM SUPER FRAME TIMING ADJUSTMENT
END

FIG.55

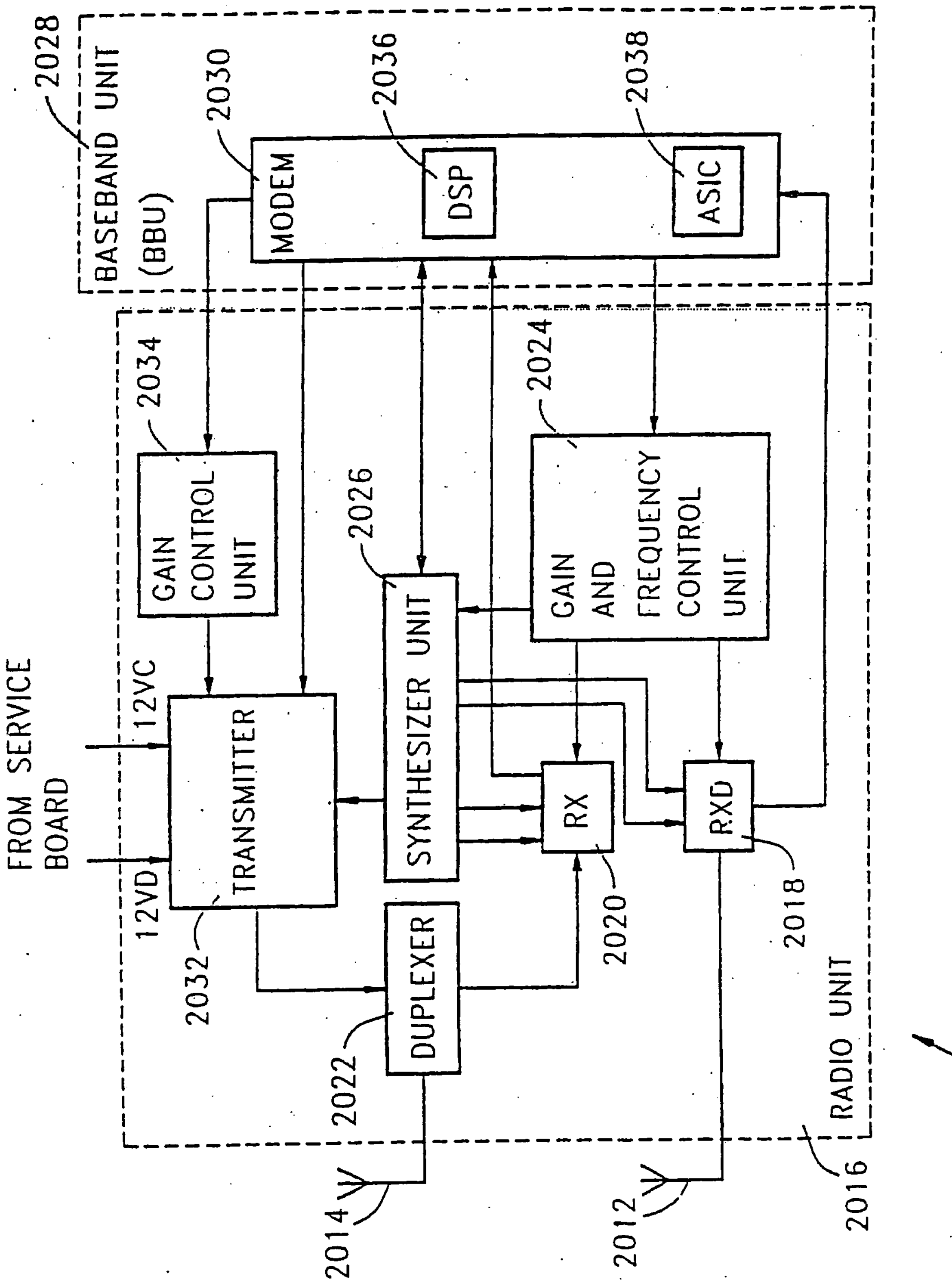

BASEBAND UNIT
(BBU)
2028
2030
MODEM
2036
DSP
2038
ASIC
2034
GAIN
CONTROL
UNIT
2024
GAIN
AND
FREQUENCY
CONTROL
UNIT
2026
SYNTHESIZER UNIT
FROM SERVICE
BOARD
12VC
12VD
TRANSMITTER
2032
DUPLEXER
2022
RX
2020
RXD
2018
RADIO UNIT
2016
2014
2012
2010

FIG. 56

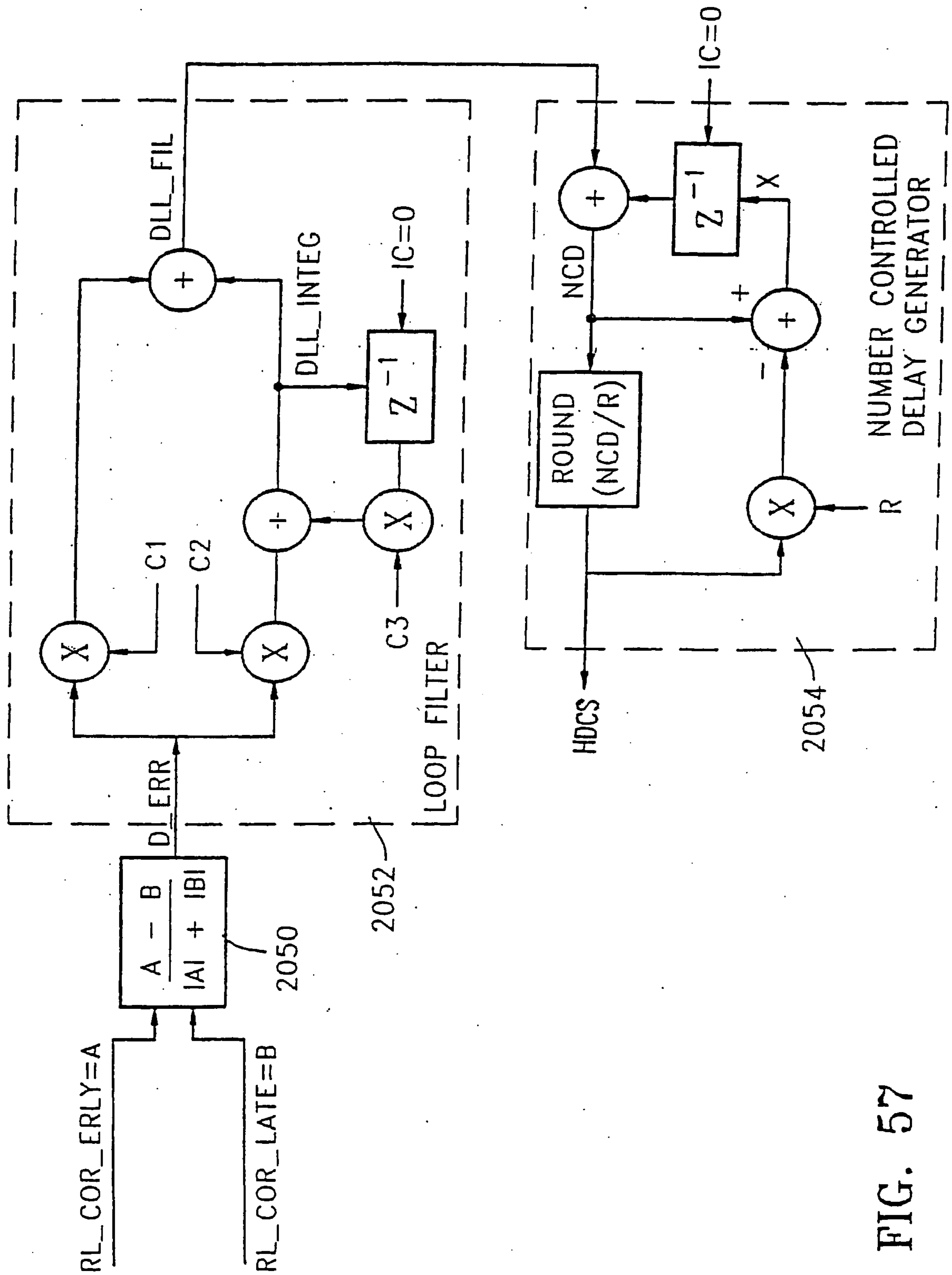
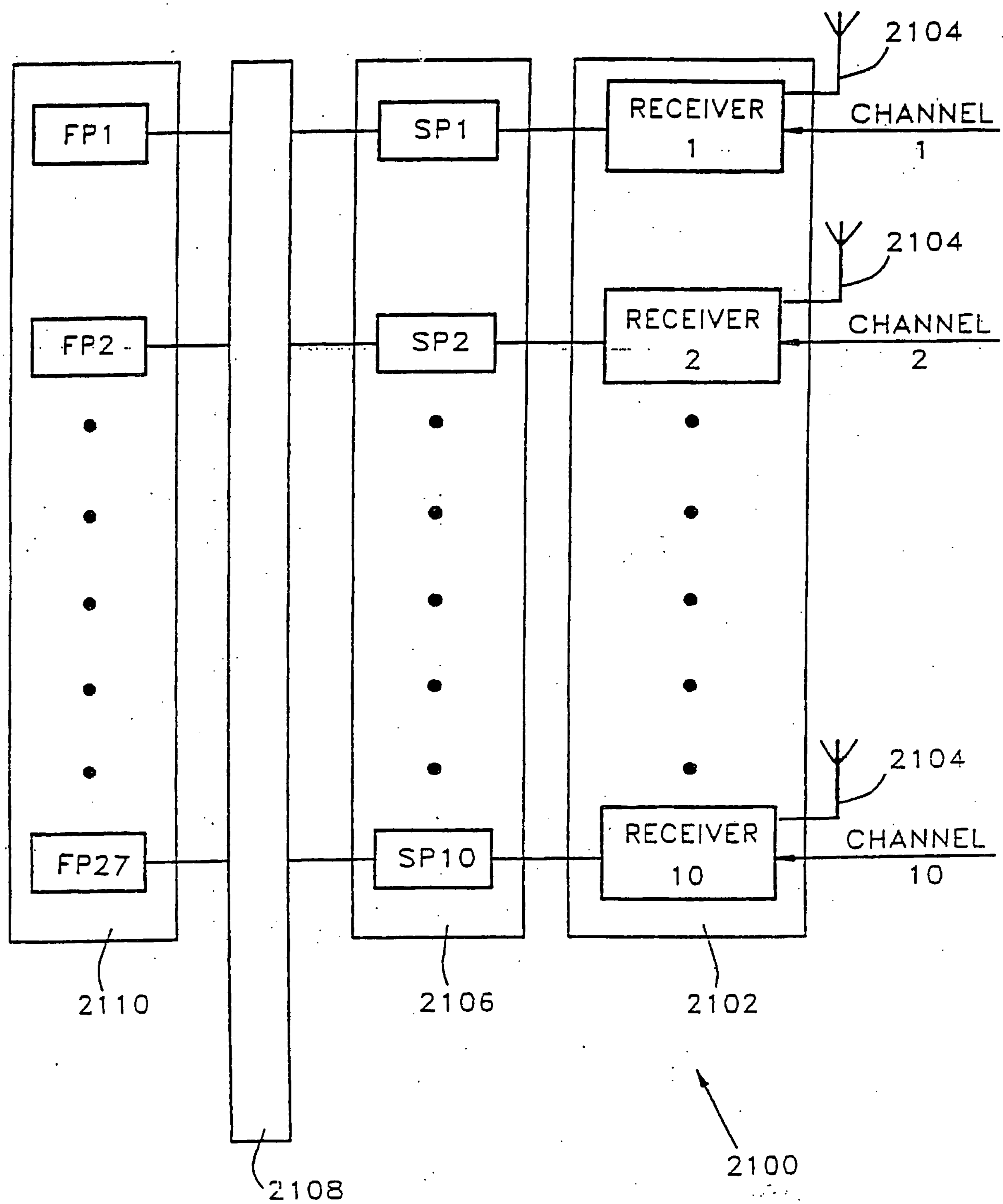

RL_COR_ERLY=A
RL_COR_LATE=B
A − B
|A| + |B|
2050
DL_ERR
C1
C2
C3
DLL_INTEG
Z⁻¹
IC=0
DLL_FIL
LOOP FILTER
2052
NCD
Z⁻¹
IC=0
X
+
−
ROUND (NCD/R)
R
HDCS
NUMBER CONTROLLED DELAY GENERATOR
2054

FIG. 57



2104
2104
2104
FP1
FP2
FP27
SP1
SP2
SP10
RECEIVER 1
RECEIVER 2
RECEIVER 10
CHANNEL 1
CHANNEL 2
CHANNEL 10
2110
2106
2102
2108
2100

FIG. 58

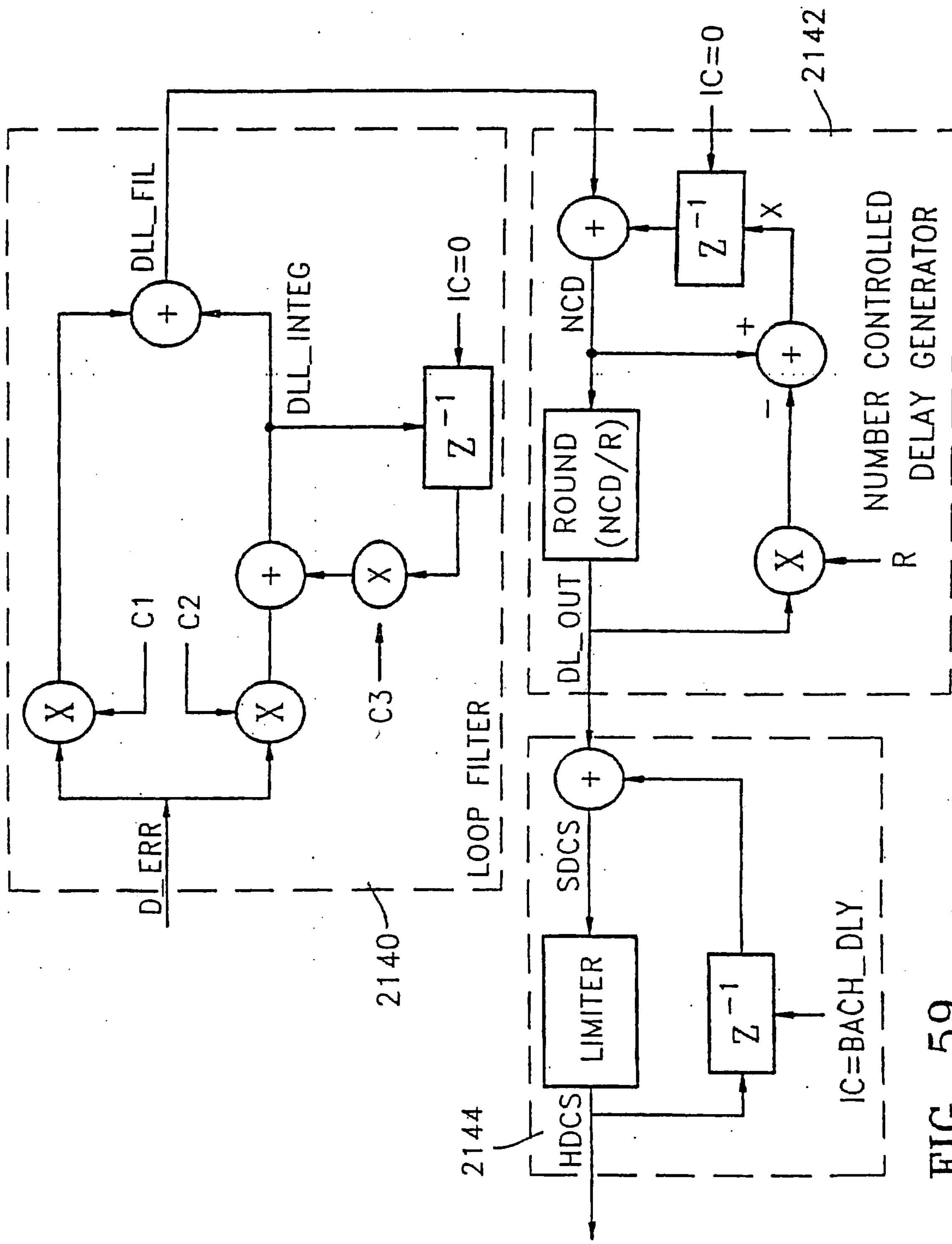

DLL_FIL
DLL_INTEG
IC=0
Z^-1
C1
C2
C3
X
+
DL_ERR
LOOP FILTER
2140
NCD
ROUND (NCD/R)
DL_OUT
R
NUMBER CONTROLLED DELAY GENERATOR
2142
SDCS
LIMITER
HDCS
IC=BACH_DLY
2144

FIG. 59

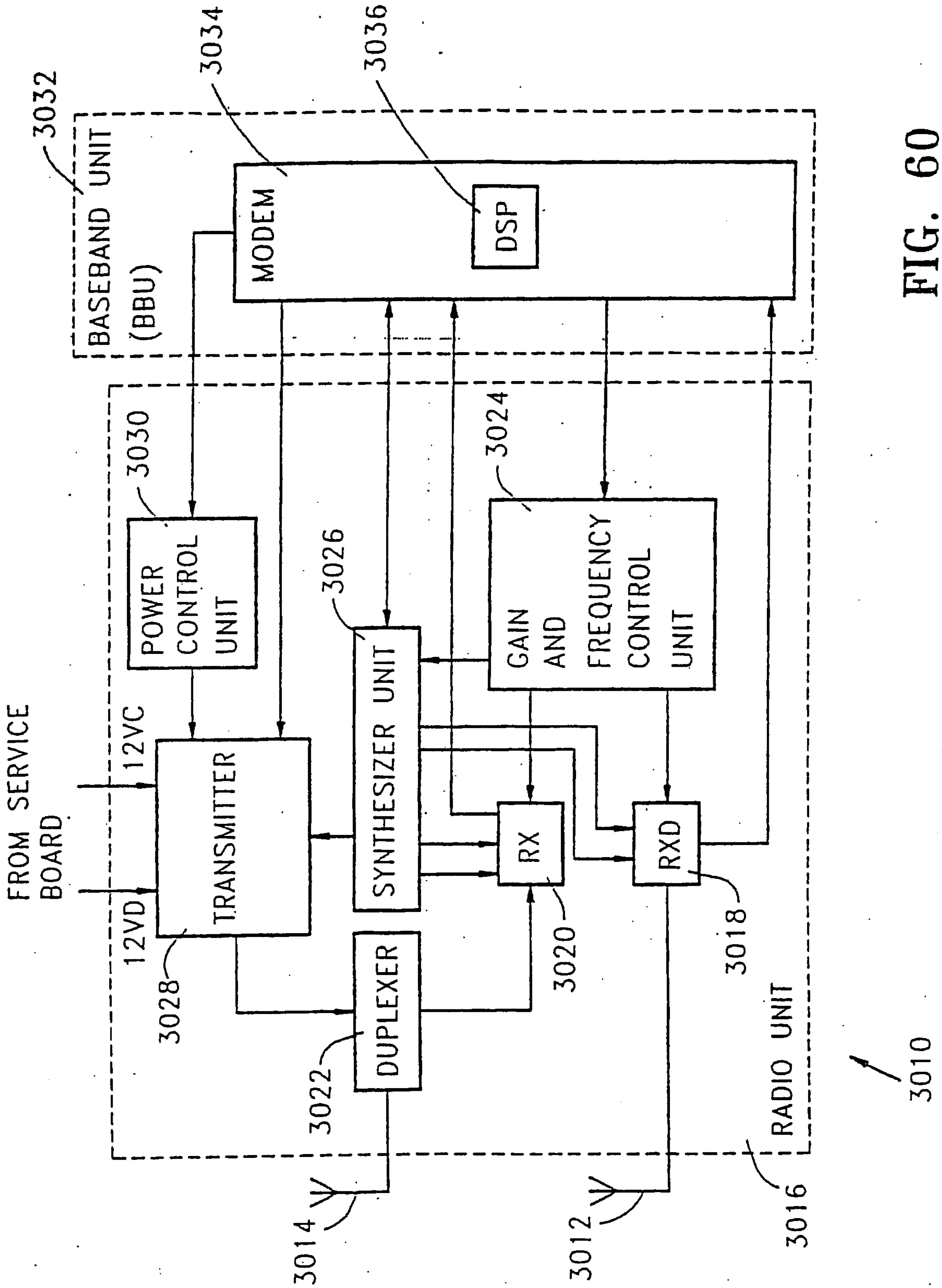

BASEBAND UNIT (BBU)
3032
3034
3036
MODEM
DSP
3030
POWER CONTROL UNIT
3024
GAIN AND FREQUENCY CONTROL UNIT
3026
SYNTHESIZER UNIT
FROM SERVICE BOARD
12VC
12VD
TRANSMITTER
3028
RX
RXD
3020
3018
DUPLEXER
3022
RADIO UNIT
3010
3014
3012
3016

FIG. 60

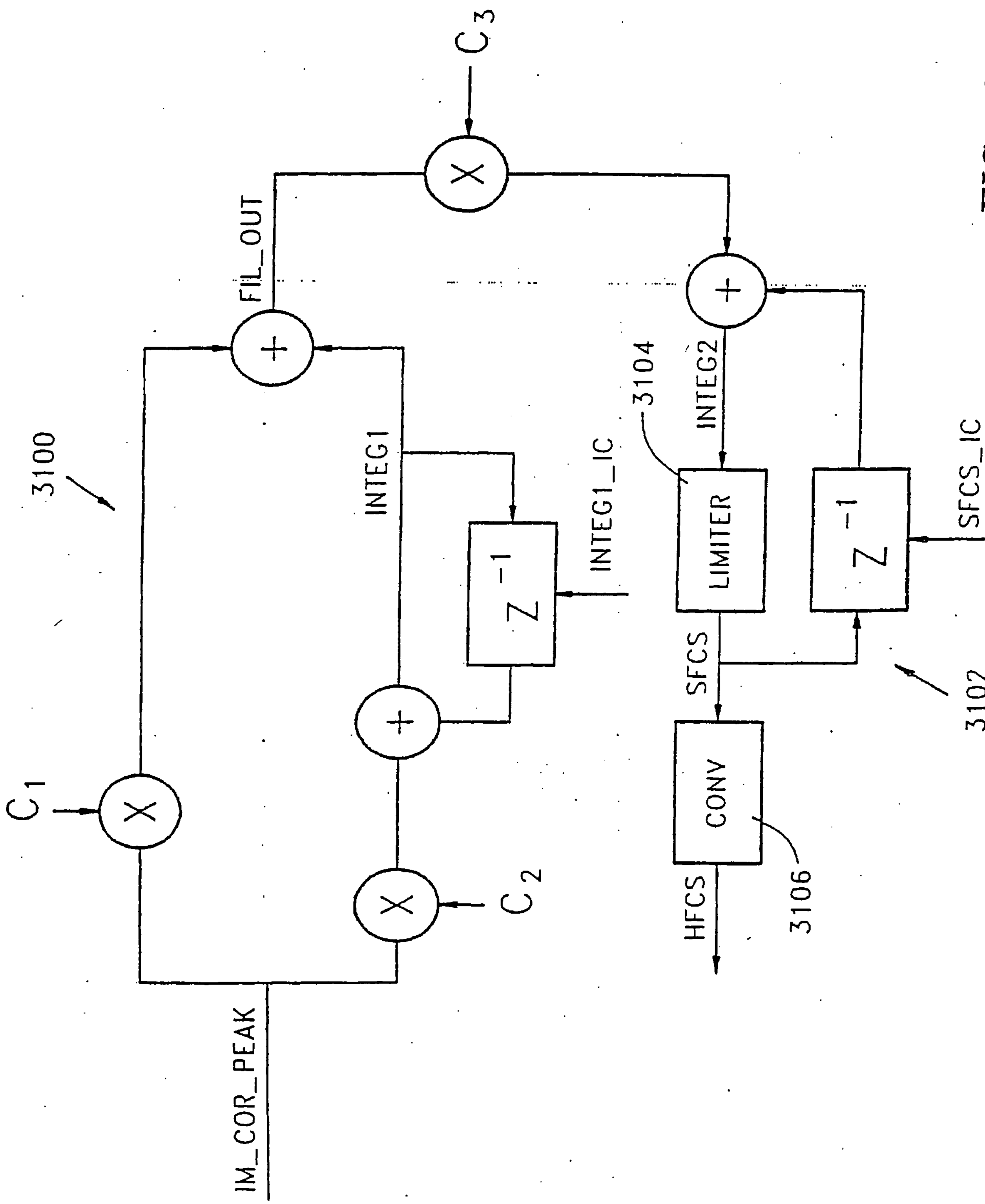

3100
IM_COR_PEAK
C1
C2
C3
FIL_OUT
INTEG1
INTEG1_IC
Z^-1
3104
INTEG2
LIMITER
SFCS
SFCS_IC
Z^-1
3102
CONV
3106
HFCS

FIG. 61

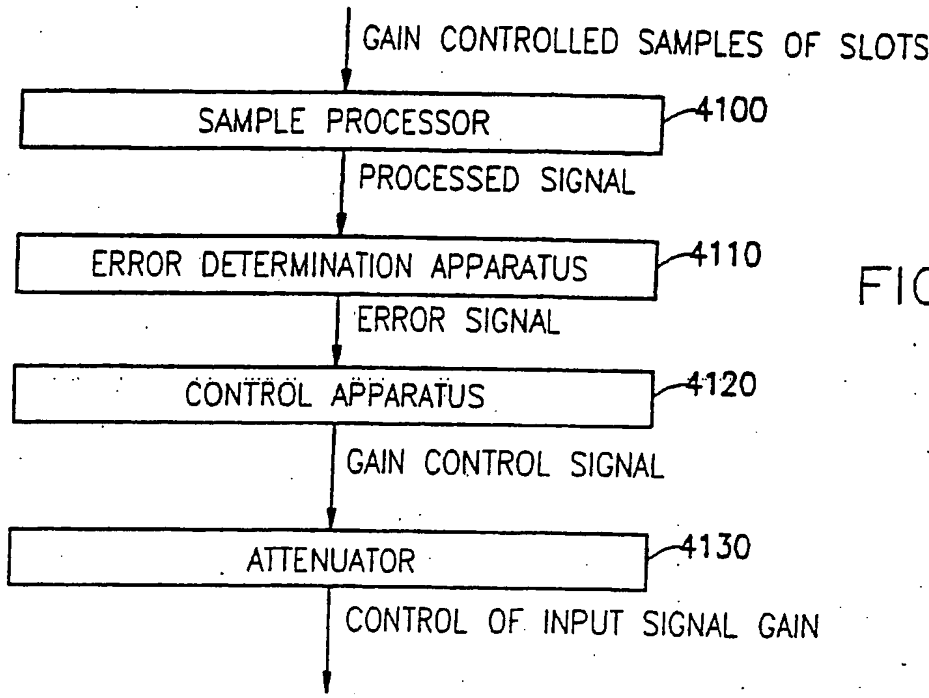

GAIN CONTROLLED SAMPLES OF SLOTS
SAMPLE PROCESSOR
4100
PROCESSED SIGNAL
ERROR DETERMINATION APPARATUS
4110
ERROR SIGNAL
CONTROL APPARATUS
4120
GAIN CONTROL SIGNAL
ATTENUATOR
4130
CONTROL OF INPUT SIGNAL GAIN

FIG. 62

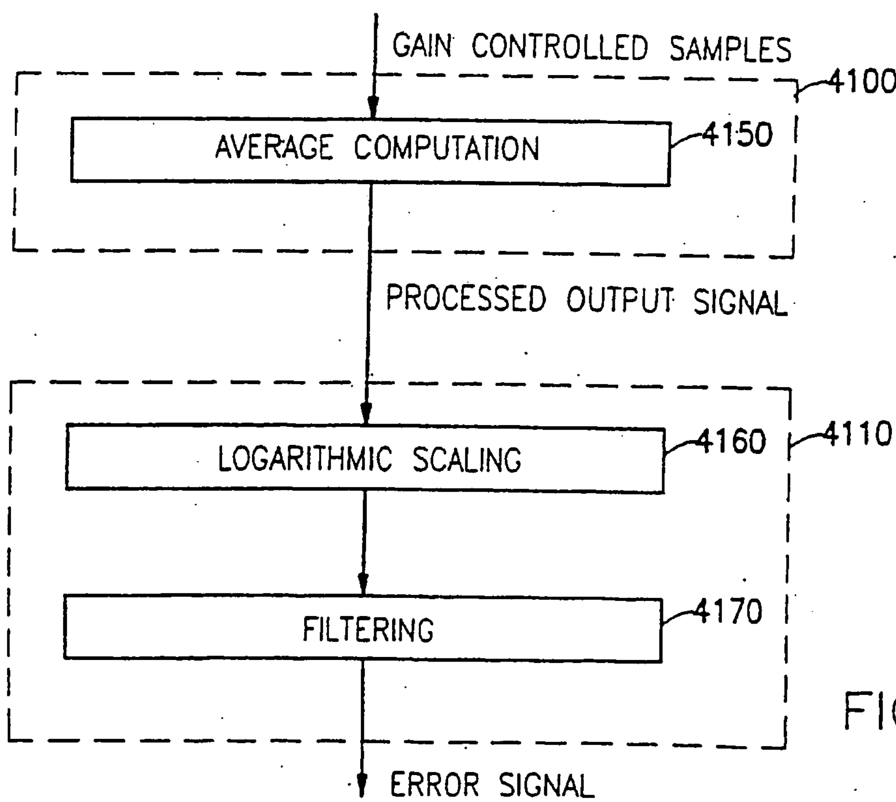
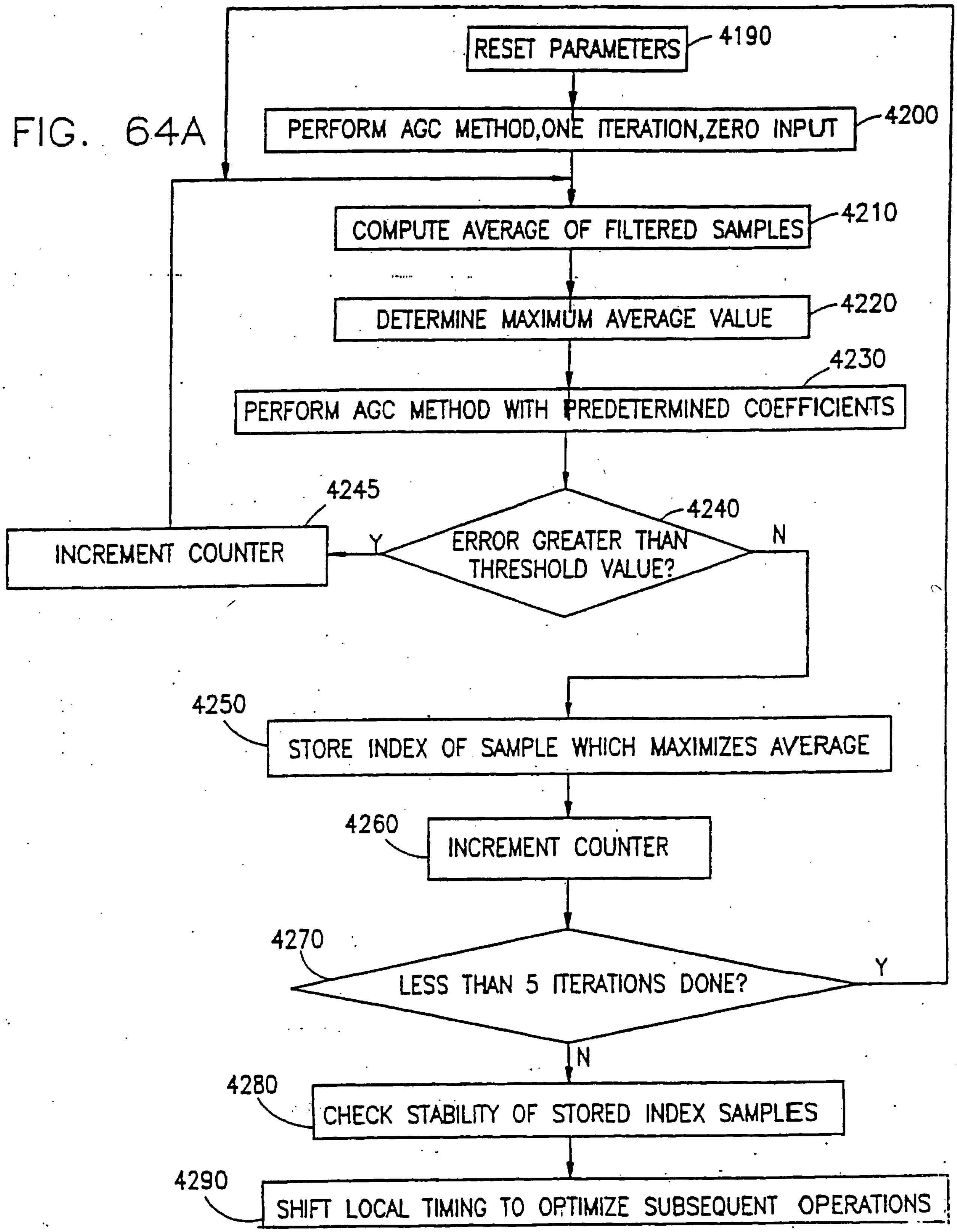

GAIN CONTROLLED SAMPLES
4100
AVERAGE COMPUTATION
4150
PROCESSED OUTPUT SIGNAL
LOGARITHMIC SCALING
4160
4110
FILTERING
4170
ERROR SIGNAL

FIG. 63

FIG. 64A


RESET PARAMETERS
4190
PERFORM AGC METHOD,ONE ITERATION,ZERO INPUT
4200
COMPUTE AVERAGE OF FILTERED SAMPLES
4210
DETERMINE MAXIMUM AVERAGE VALUE
4220
4230
PERFORM AGC METHOD WITH PREDETERMINED COEFFICIENTS
4245
4240
INCREMENT COUNTER
Y
ERROR GREATER THAN THRESHOLD VALUE?
N
4250
STORE INDEX OF SAMPLE WHICH MAXIMIZES AVERAGE
4260
INCREMENT COUNTER
4270
LESS THAN 5 ITERATIONS DONE?
Y
N
4280
CHECK STABILITY OF STORED INDEX SAMPLES
4290
SHIFT LOCAL TIMING TO OPTIMIZE SUBSEQUENT OPERATIONS

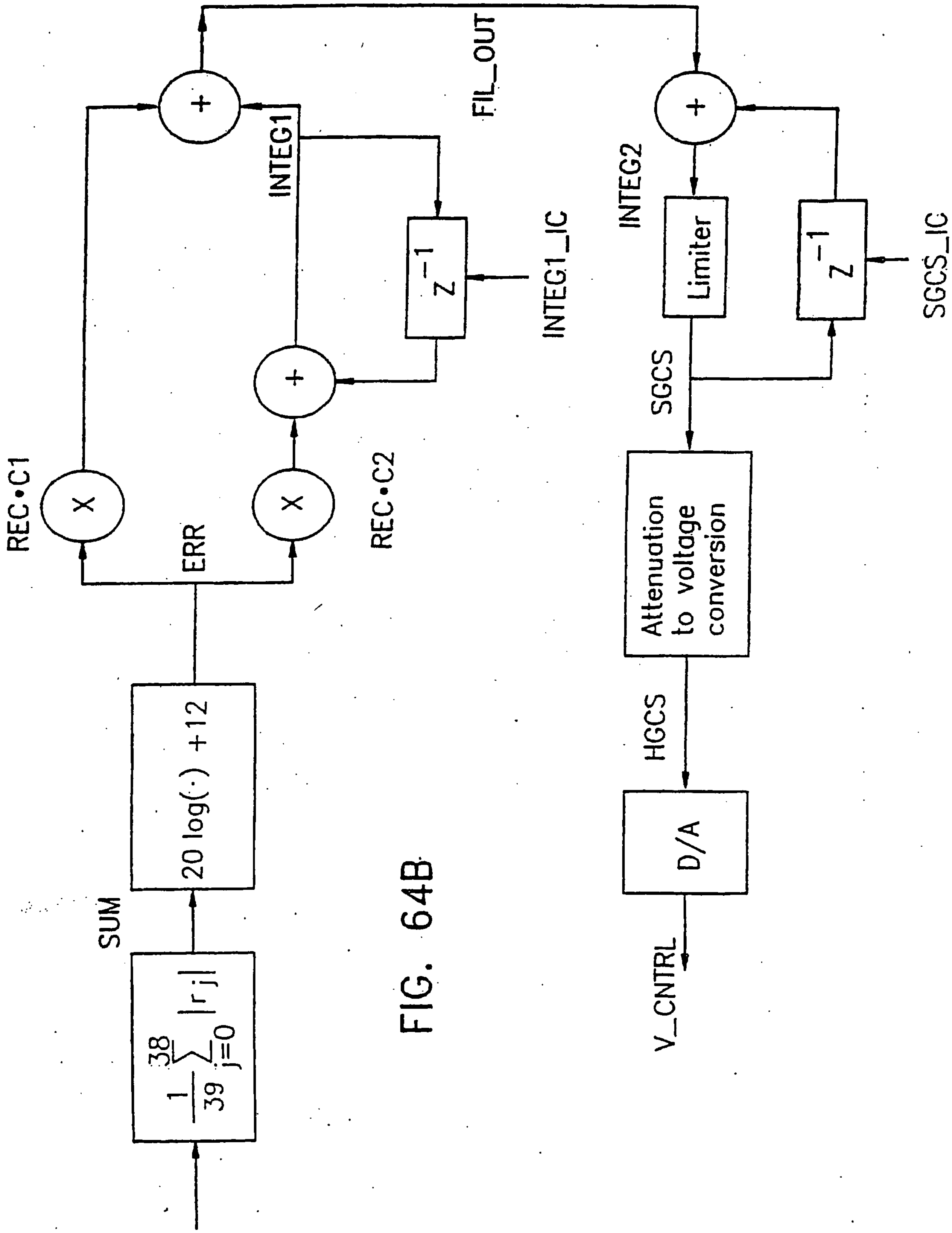
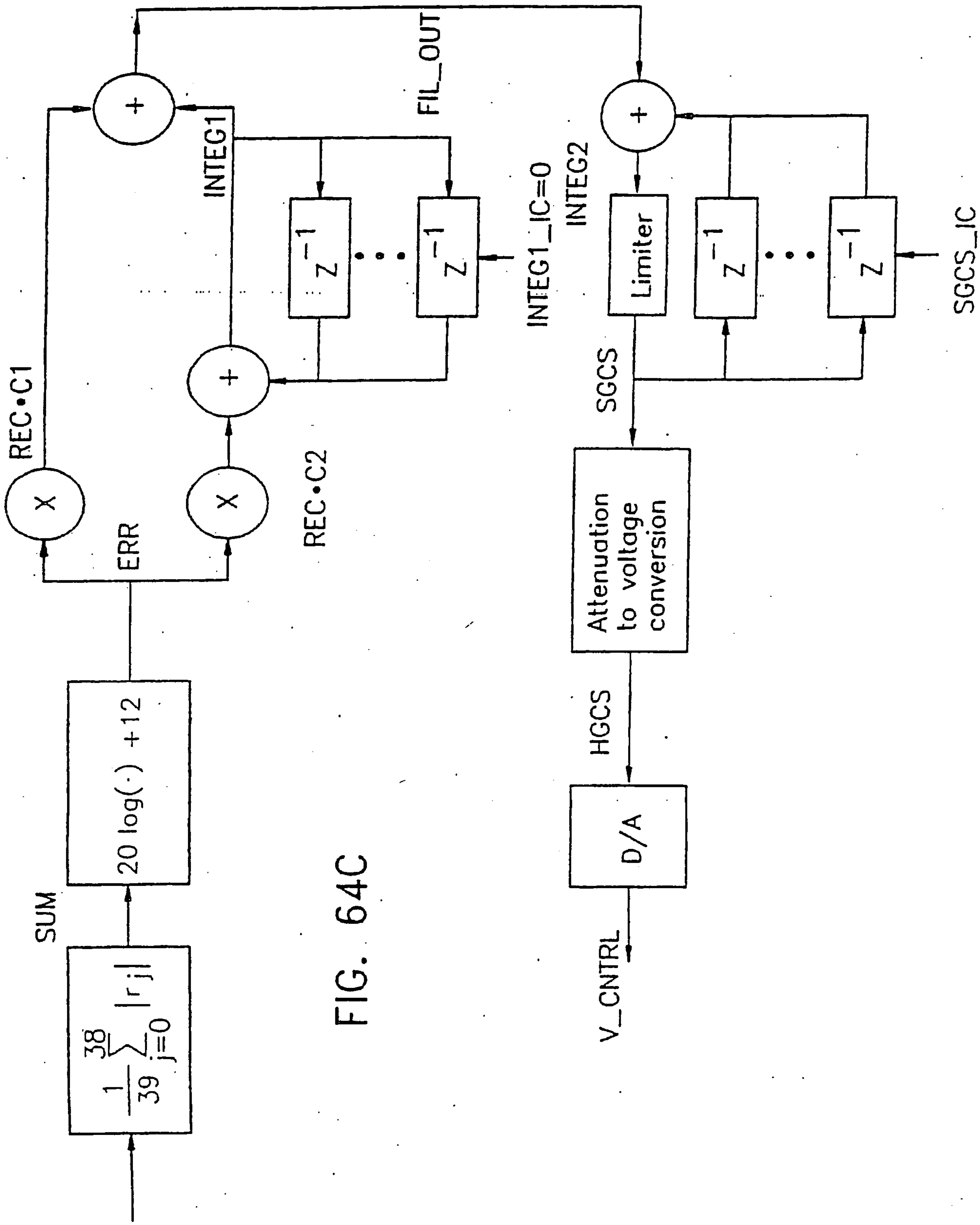

FIL_OUT
INTEG1
INTEG1_IC
INTEG2
Limiter
SGCS_IC
SGCS
REC•C1
REC•C2
ERR
Attenuation to voltage conversion
HGCS
D/A
V_CNTRL
SUM
20 log(·) +12
$\frac{1}{39}\sum_{j=0}^{38}|r_j|$
z^{-1}

FIG. 64B



SUM
20 log(·) +12
ERR
REC•C1
REC•C2
INTEG1
FIL_OUT
INTEG1_IC=0
INTEG2
Limiter
SGCS_IC
SGCS
Attenuation to voltage conversion
HGCS
D/A
V_CNTRL

FIG. 64C

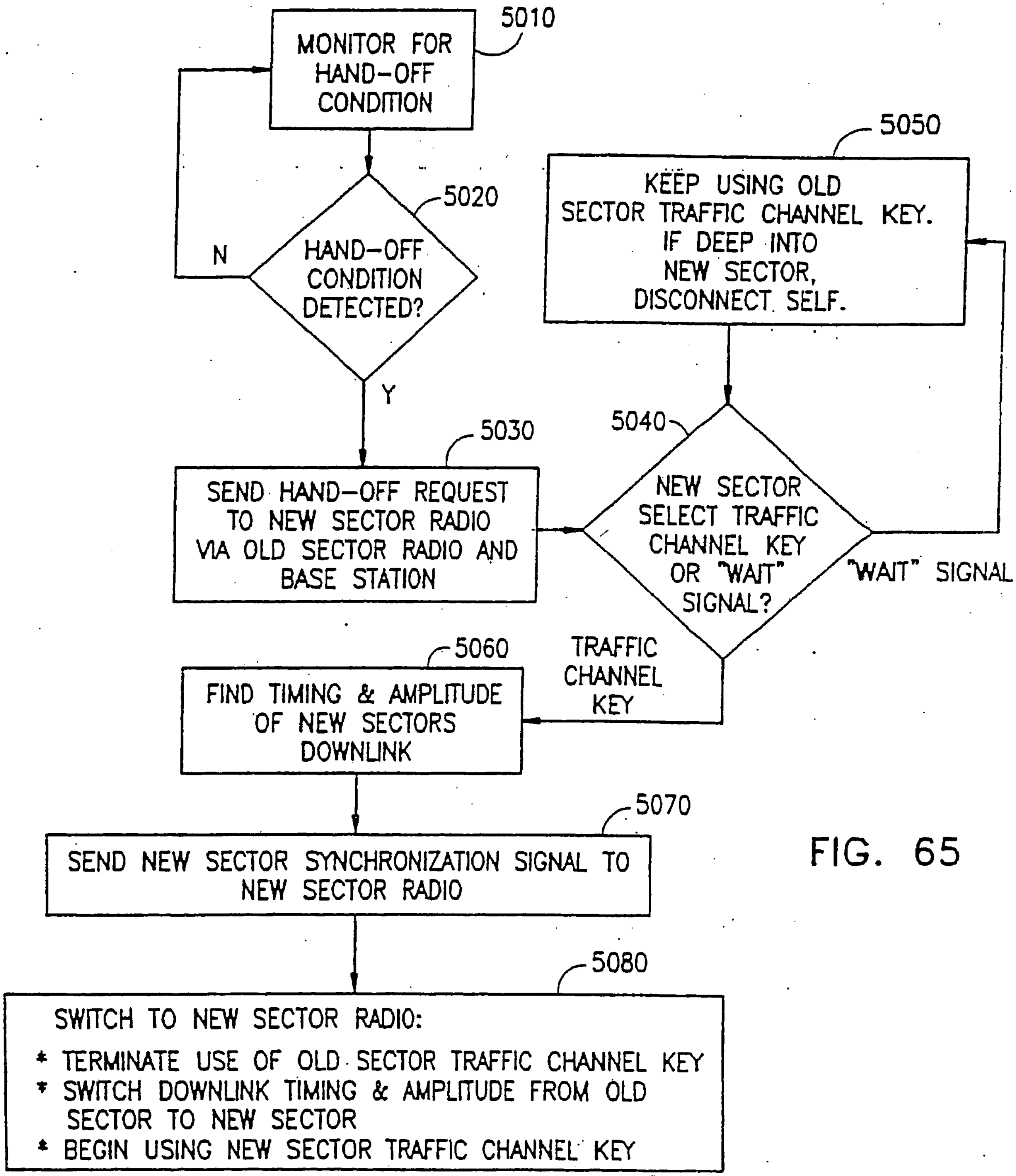

MONITOR FOR HAND-OFF CONDITION
5010
HAND-OFF CONDITION DETECTED?
5020
N
Y
SEND HAND-OFF REQUEST TO NEW SECTOR RADIO VIA OLD SECTOR RADIO AND BASE STATION
5030
KEEP USING OLD SECTOR TRAFFIC CHANNEL KEY. IF DEEP INTO NEW SECTOR, DISCONNECT SELF.
5050
NEW SECTOR SELECT TRAFFIC CHANNEL KEY OR "WAIT" SIGNAL?
5040
"WAIT" SIGNAL
TRAFFIC CHANNEL KEY
FIND TIMING & AMPLITUDE OF NEW SECTORS DOWNLINK
5060
SEND NEW SECTOR SYNCHRONIZATION SIGNAL TO NEW SECTOR RADIO
5070
SWITCH TO NEW SECTOR RADIO:
* TERMINATE USE OF OLD SECTOR TRAFFIC CHANNEL KEY
* SWITCH DOWNLINK TIMING & AMPLITUDE FROM OLD SECTOR TO NEW SECTOR
* BEGIN USING NEW SECTOR TRAFFIC CHANNEL KEY
5080

FIG. 65

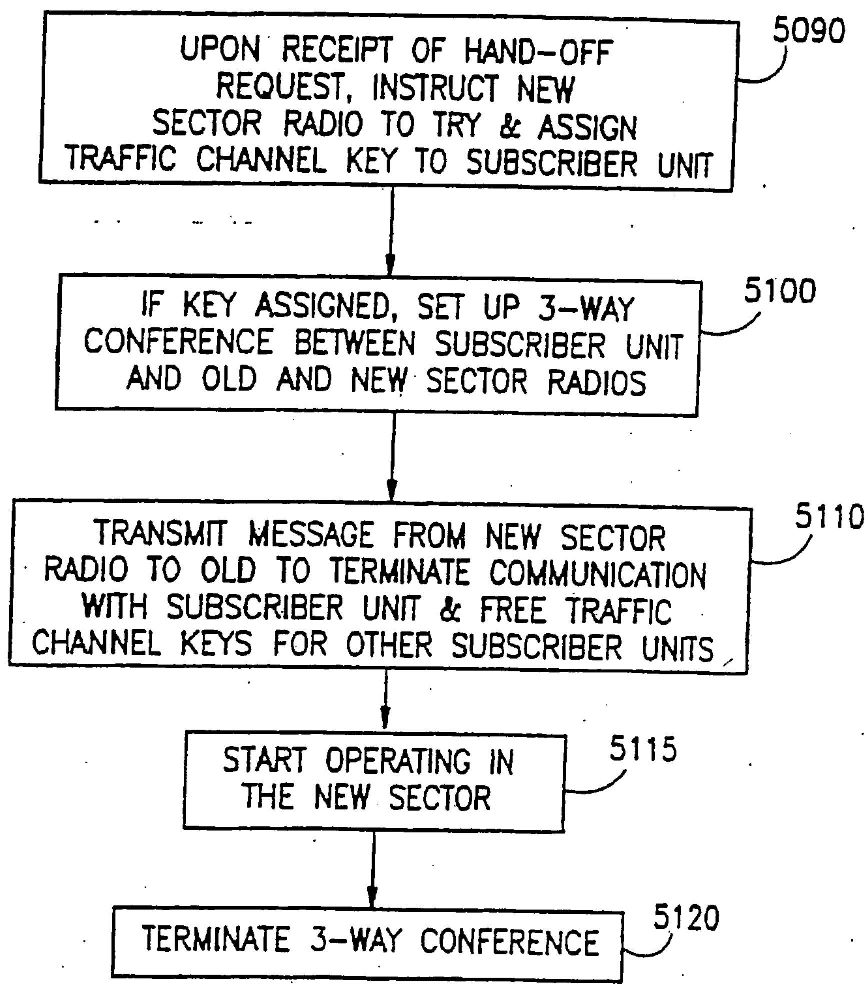

UPON RECEIPT OF HAND-OFF REQUEST, INSTRUCT NEW SECTOR RADIO TO TRY & ASSIGN TRAFFIC CHANNEL KEY TO SUBSCRIBER UNIT
5090
IF KEY ASSIGNED, SET UP 3-WAY CONFERENCE BETWEEN SUBSCRIBER UNIT AND OLD AND NEW SECTOR RADIOS
5100
TRANSMIT MESSAGE FROM NEW SECTOR RADIO TO OLD TO TERMINATE COMMUNICATION WITH SUBSCRIBER UNIT & FREE TRAFFIC CHANNEL KEYS FOR OTHER SUBSCRIBER UNITS
5110
START OPERATING IN THE NEW SECTOR
5115
TERMINATE 3-WAY CONFERENCE
5120

FIG. 66

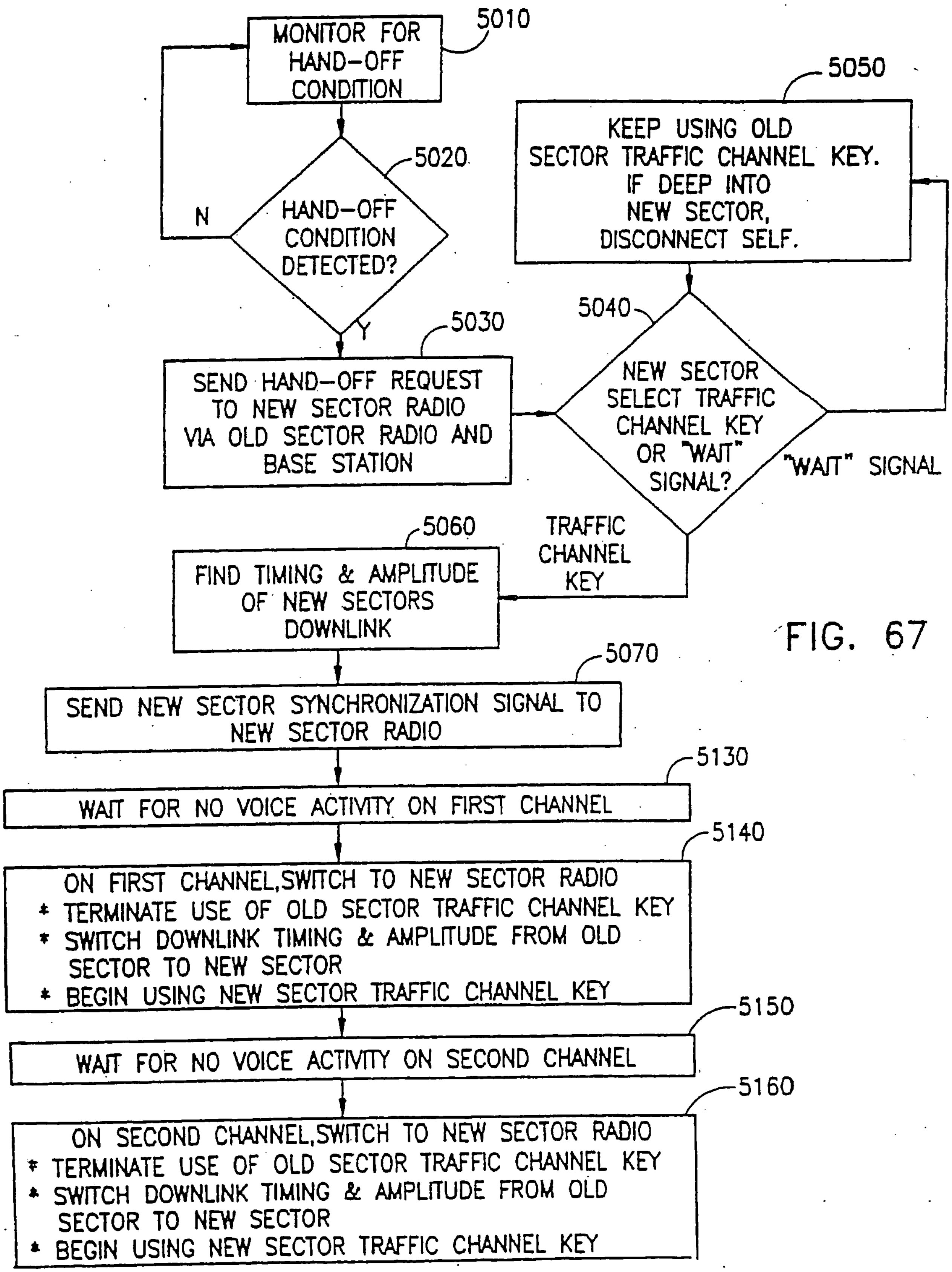
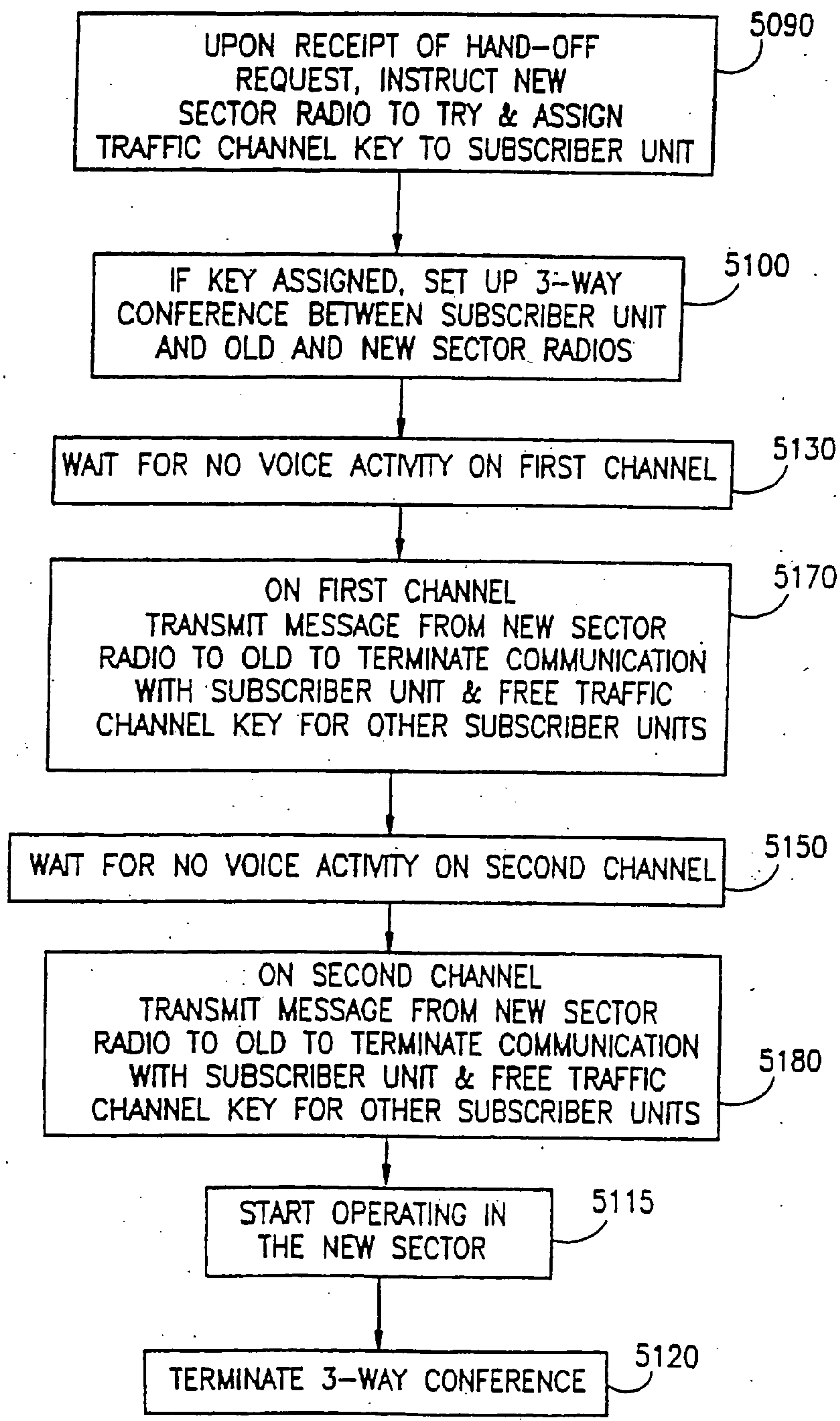

MONITOR FOR HAND-OFF CONDITION
5010
HAND-OFF CONDITION DETECTED?
5020
N
Y
SEND HAND-OFF REQUEST TO NEW SECTOR RADIO VIA OLD SECTOR RADIO AND BASE STATION
5030
KEEP USING OLD SECTOR TRAFFIC CHANNEL KEY. IF DEEP INTO NEW SECTOR, DISCONNECT SELF.
5050
NEW SECTOR SELECT TRAFFIC CHANNEL KEY OR "WAIT" SIGNAL?
5040
"WAIT" SIGNAL
TRAFFIC CHANNEL KEY
FIND TIMING & AMPLITUDE OF NEW SECTORS DOWNLINK
5060
SEND NEW SECTOR SYNCHRONIZATION SIGNAL TO NEW SECTOR RADIO
5070
WAIT FOR NO VOICE ACTIVITY ON FIRST CHANNEL
5130
ON FIRST CHANNEL,SWITCH TO NEW SECTOR RADIO
* TERMINATE USE OF OLD SECTOR TRAFFIC CHANNEL KEY
* SWITCH DOWNLINK TIMING & AMPLITUDE FROM OLD SECTOR TO NEW SECTOR
* BEGIN USING NEW SECTOR TRAFFIC CHANNEL KEY
5140
WAIT FOR NO VOICE ACTIVITY ON SECOND CHANNEL
5150
ON SECOND CHANNEL,SWITCH TO NEW SECTOR RADIO
* TERMINATE USE OF OLD SECTOR TRAFFIC CHANNEL KEY
* SWITCH DOWNLINK TIMING & AMPLITUDE FROM OLD SECTOR TO NEW SECTOR
* BEGIN USING NEW SECTOR TRAFFIC CHANNEL KEY
5160

FIG. 67

FIG. 68


UPON RECEIPT OF HAND-OFF REQUEST, INSTRUCT NEW SECTOR RADIO TO TRY & ASSIGN TRAFFIC CHANNEL KEY TO SUBSCRIBER UNIT
5090
IF KEY ASSIGNED, SET UP 3-WAY CONFERENCE BETWEEN SUBSCRIBER UNIT AND OLD AND NEW SECTOR RADIOS
5100
WAIT FOR NO VOICE ACTIVITY ON FIRST CHANNEL
5130
ON FIRST CHANNEL TRANSMIT MESSAGE FROM NEW SECTOR RADIO TO OLD TO TERMINATE COMMUNICATION WITH SUBSCRIBER UNIT & FREE TRAFFIC CHANNEL KEY FOR OTHER SUBSCRIBER UNITS
5170
WAIT FOR NO VOICE ACTIVITY ON SECOND CHANNEL
5150
ON SECOND CHANNEL TRANSMIT MESSAGE FROM NEW SECTOR RADIO TO OLD TO TERMINATE COMMUNICATION WITH SUBSCRIBER UNIT & FREE TRAFFIC CHANNEL KEY FOR OTHER SUBSCRIBER UNITS
5180
START OPERATING IN THE NEW SECTOR
5115
TERMINATE 3-WAY CONFERENCE
5120

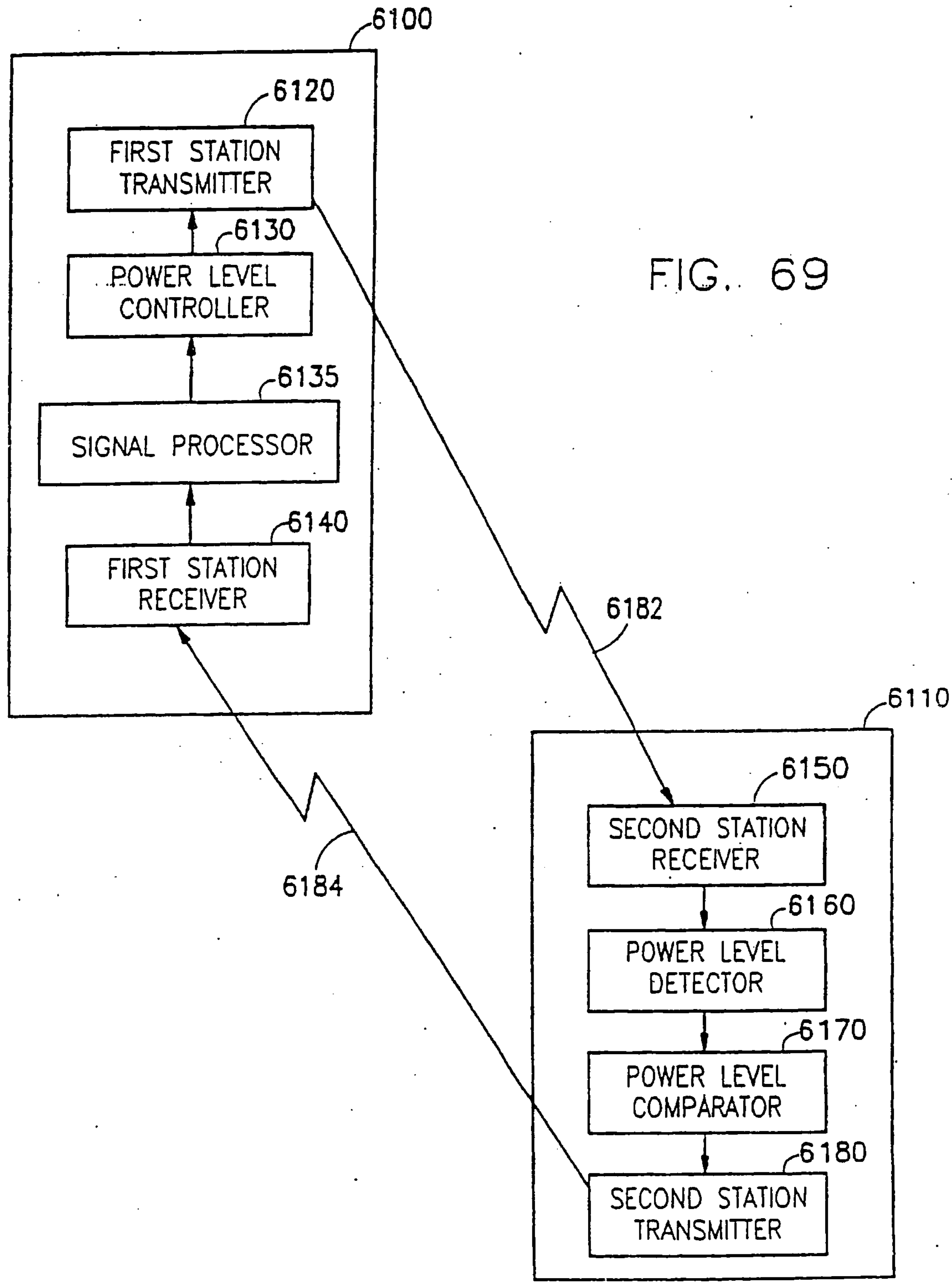

6100
6120
FIRST STATION TRANSMITTER
6130
POWER LEVEL CONTROLLER
6135
SIGNAL PROCESSOR
6140
FIRST STATION RECEIVER
6182
6184
6110
6150
SECOND STATION RECEIVER
6160
POWER LEVEL DETECTOR
6170
POWER LEVEL COMPARATOR
6180
SECOND STATION TRANSMITTER

FIG. 69

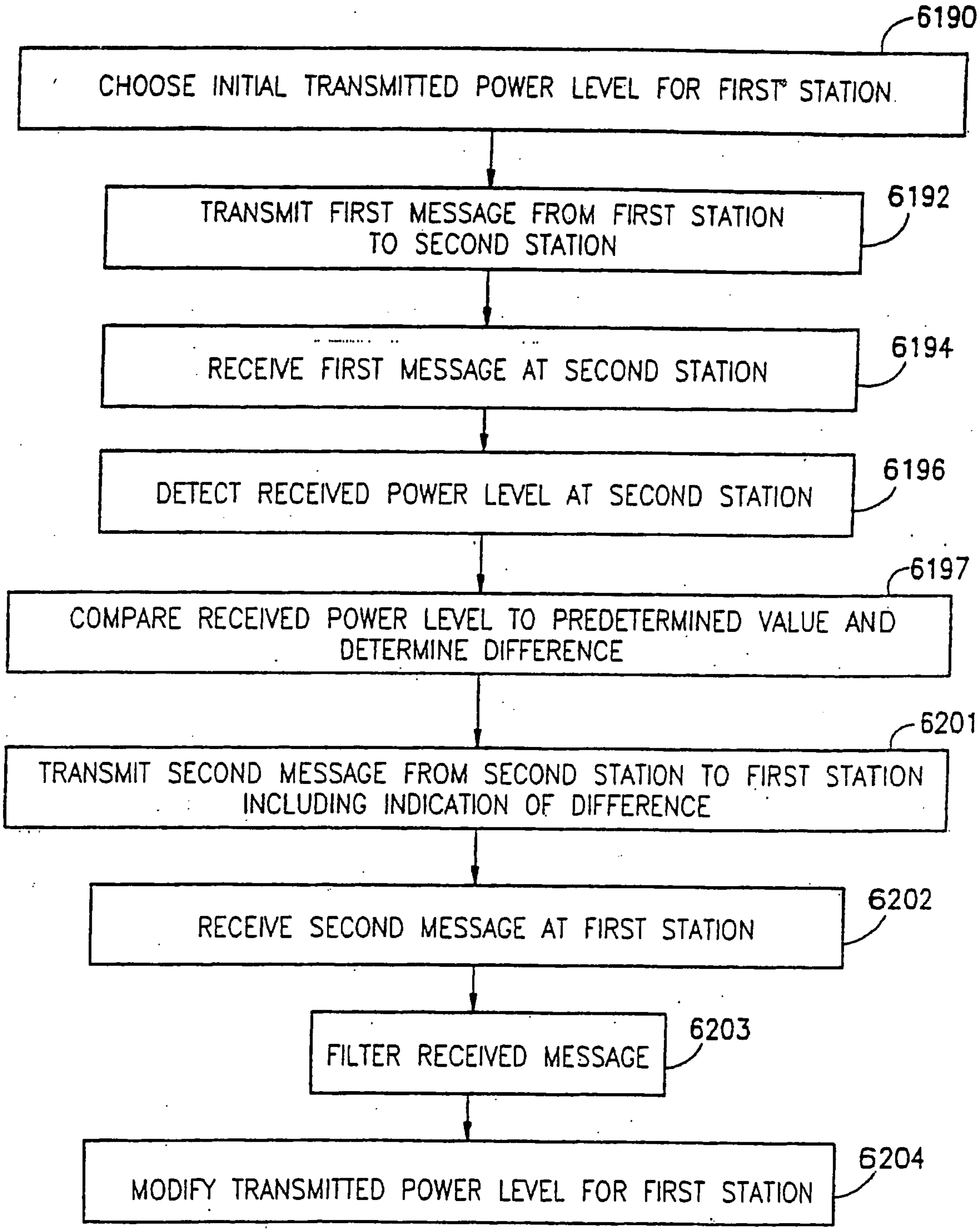

6190
CHOOSE INITIAL TRANSMITTED POWER LEVEL FOR FIRST STATION
6192
TRANSMIT FIRST MESSAGE FROM FIRST STATION TO SECOND STATION
6194
RECEIVE FIRST MESSAGE AT SECOND STATION
6196
DETECT RECEIVED POWER LEVEL AT SECOND STATION
6197
COMPARE RECEIVED POWER LEVEL TO PREDETERMINED VALUE AND DETERMINE DIFFERENCE
6201
TRANSMIT SECOND MESSAGE FROM SECOND STATION TO FIRST STATION INCLUDING INDICATION OF DIFFERENCE
6202
RECEIVE SECOND MESSAGE AT FIRST STATION
6203
FILTER RECEIVED MESSAGE
6204
MODIFY TRANSMITTED POWER LEVEL FOR FIRST STATION

FIG. 70A

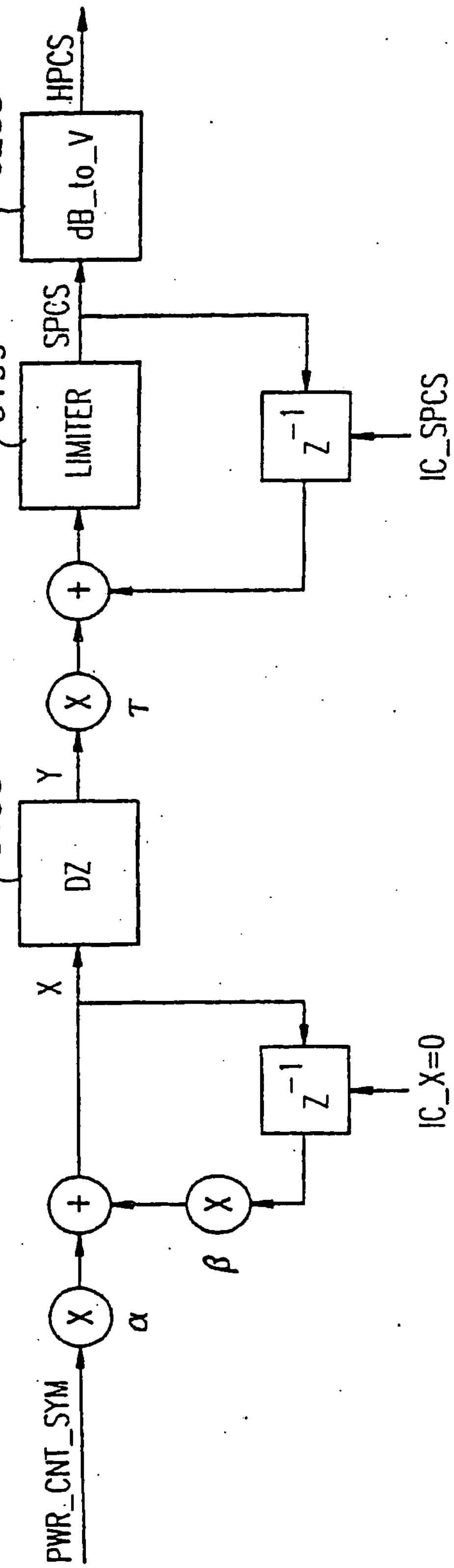

PWR_CNT_SYM
α
β
IC_X=0
X
6198
DZ
Y
τ
6199
LIMITER
IC_SPCS
SPCS
6200
dB_to_V
HPCS

FIG. 70B

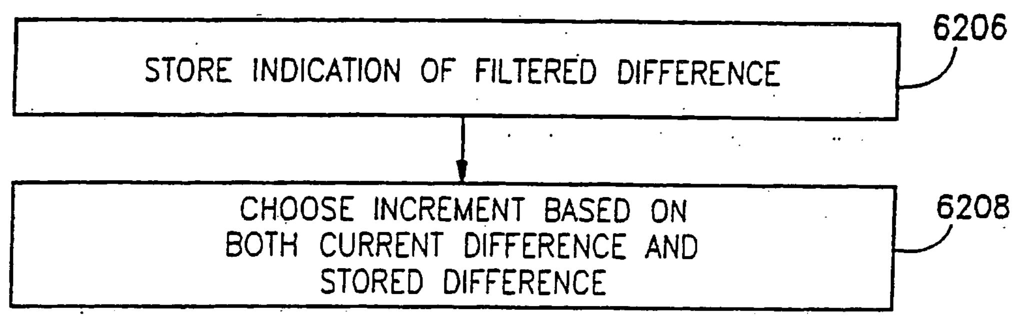

STORE INDICATION OF FILTERED DIFFERENCE
6206
CHOOSE INCREMENT BASED ON BOTH CURRENT DIFFERENCE AND STORED DIFFERENCE
6208

FIG. 71

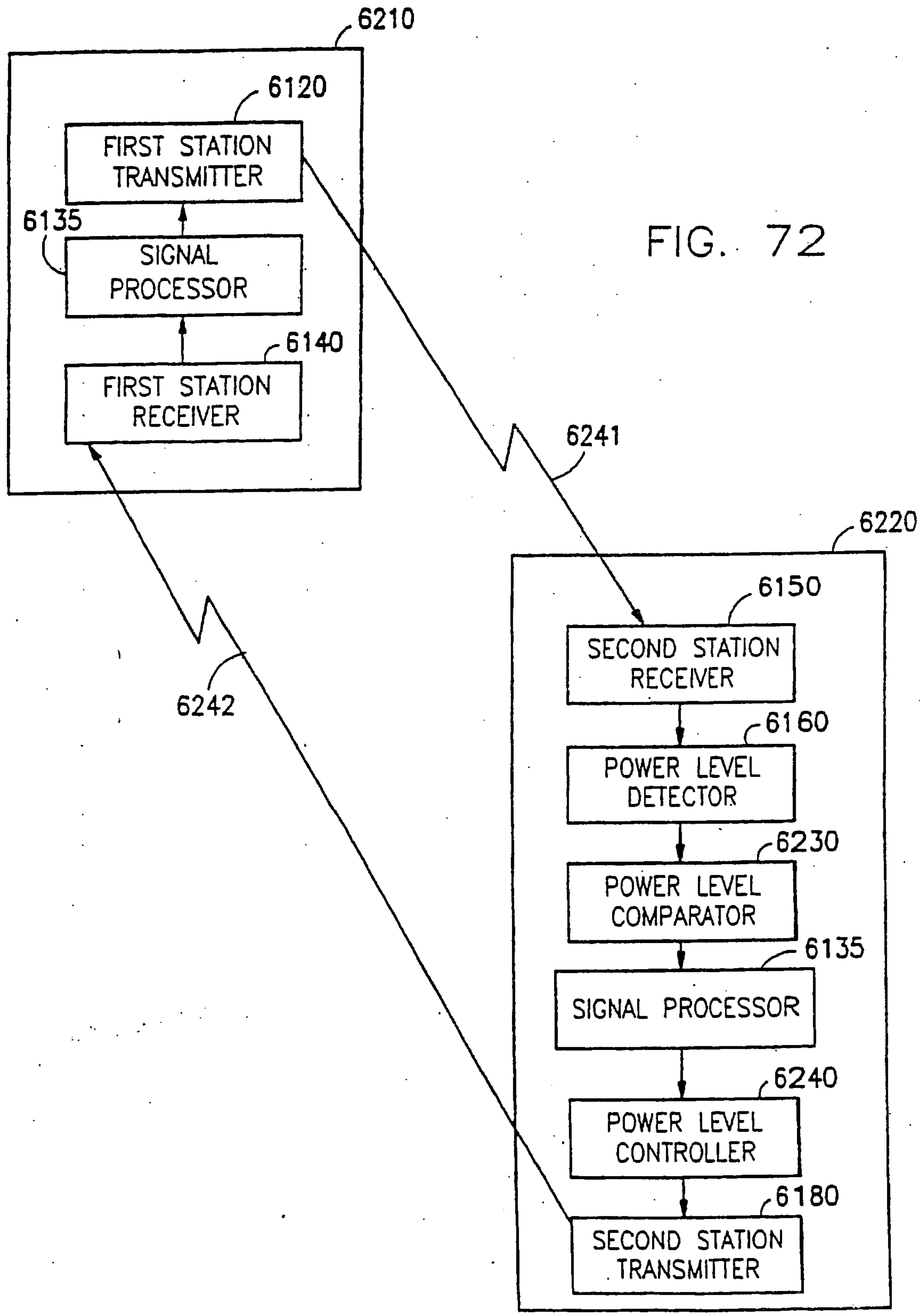

6210
6120
FIRST STATION TRANSMITTER
6135
SIGNAL PROCESSOR
6140
FIRST STATION RECEIVER
6241
6242
6220
6150
SECOND STATION RECEIVER
6160
POWER LEVEL DETECTOR
6230
POWER LEVEL COMPARATOR
6135
SIGNAL PROCESSOR
6240
POWER LEVEL CONTROLLER
6180
SECOND STATION TRANSMITTER

FIG. 72

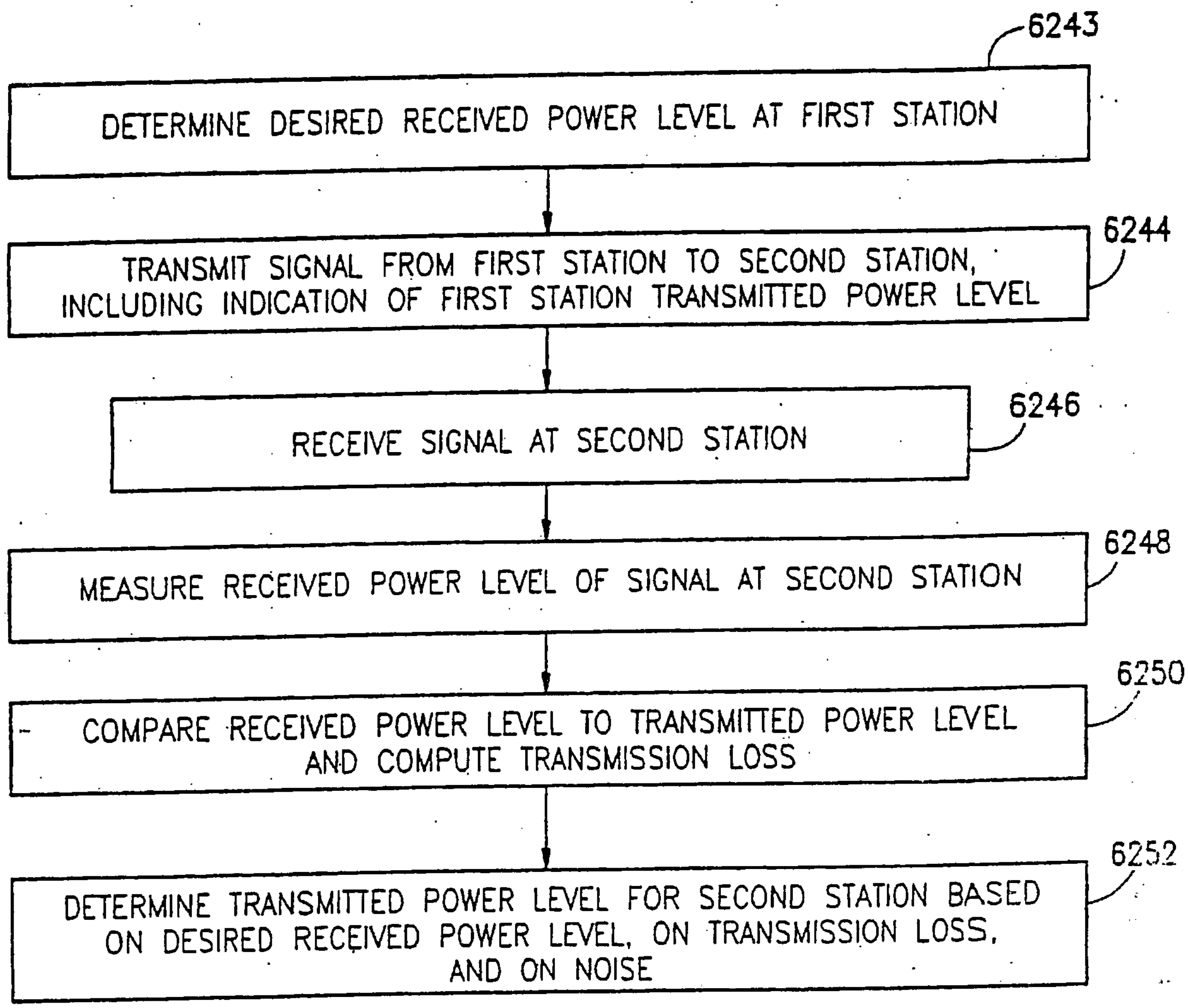

6243
DETERMINE DESIRED RECEIVED POWER LEVEL AT FIRST STATION
6244
TRANSMIT SIGNAL FROM FIRST STATION TO SECOND STATION, INCLUDING INDICATION OF FIRST STATION TRANSMITTED POWER LEVEL
6246
RECEIVE SIGNAL AT SECOND STATION
6248
MEASURE RECEIVED POWER LEVEL OF SIGNAL AT SECOND STATION
6250
COMPARE RECEIVED POWER LEVEL TO TRANSMITTED POWER LEVEL AND COMPUTE TRANSMISSION LOSS
6252
DETERMINE TRANSMITTED POWER LEVEL FOR SECOND STATION BASED ON DESIRED RECEIVED POWER LEVEL, ON TRANSMISSION LOSS, AND ON NOISE

FIG. 73

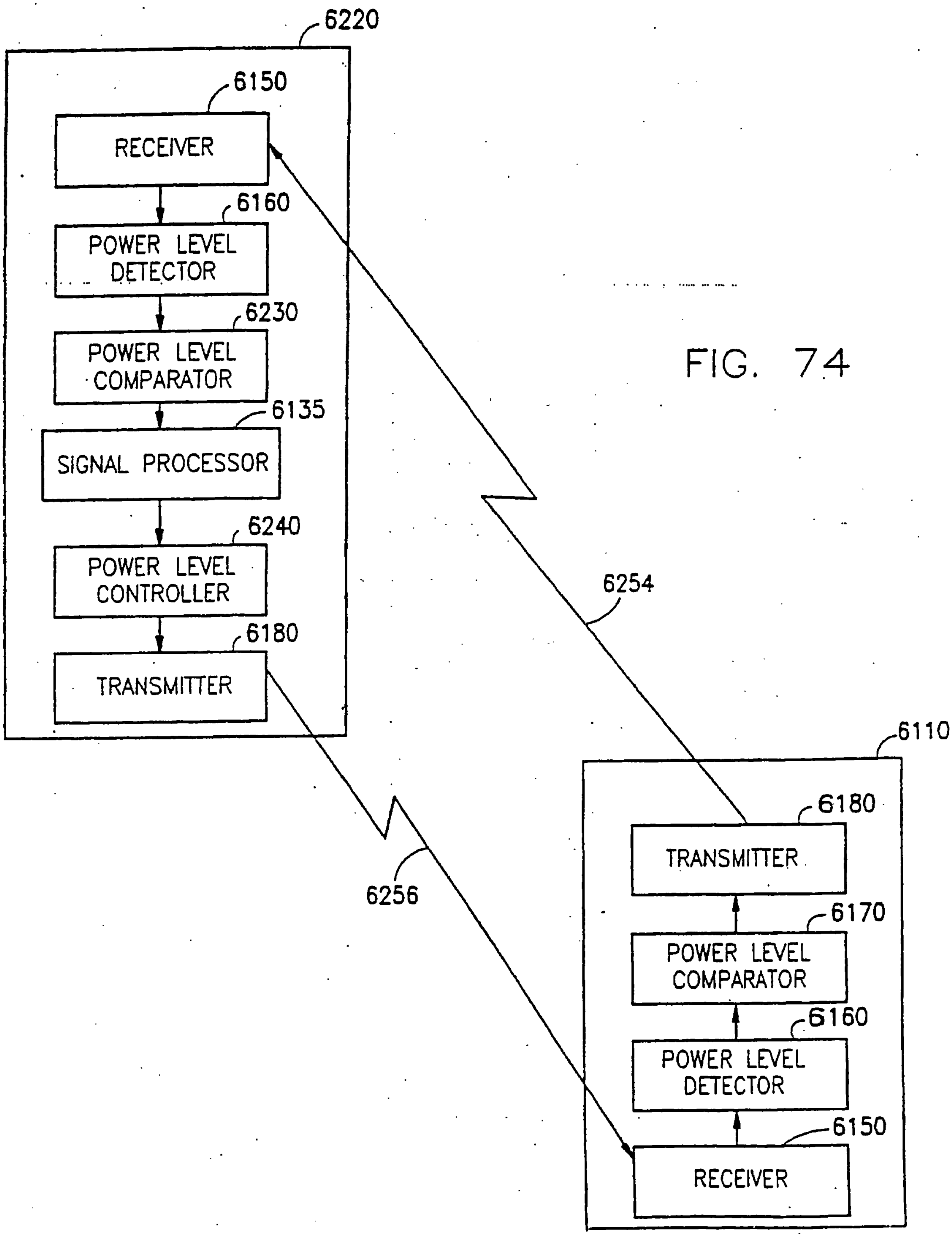

6220
6150
RECEIVER
6160
POWER LEVEL DETECTOR
6230
POWER LEVEL COMPARATOR
6135
SIGNAL PROCESSOR
6240
POWER LEVEL CONTROLLER
6180
TRANSMITTER
6254
6256
6110
6180
TRANSMITTER
6170
POWER LEVEL COMPARATOR
6160
POWER LEVEL DETECTOR
6150
RECEIVER

FIG. 74

FIG. 75

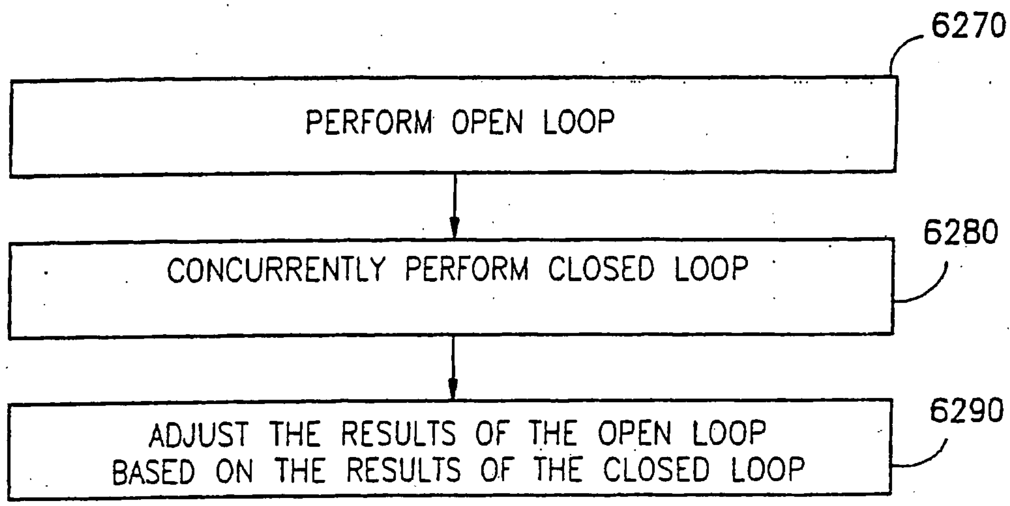
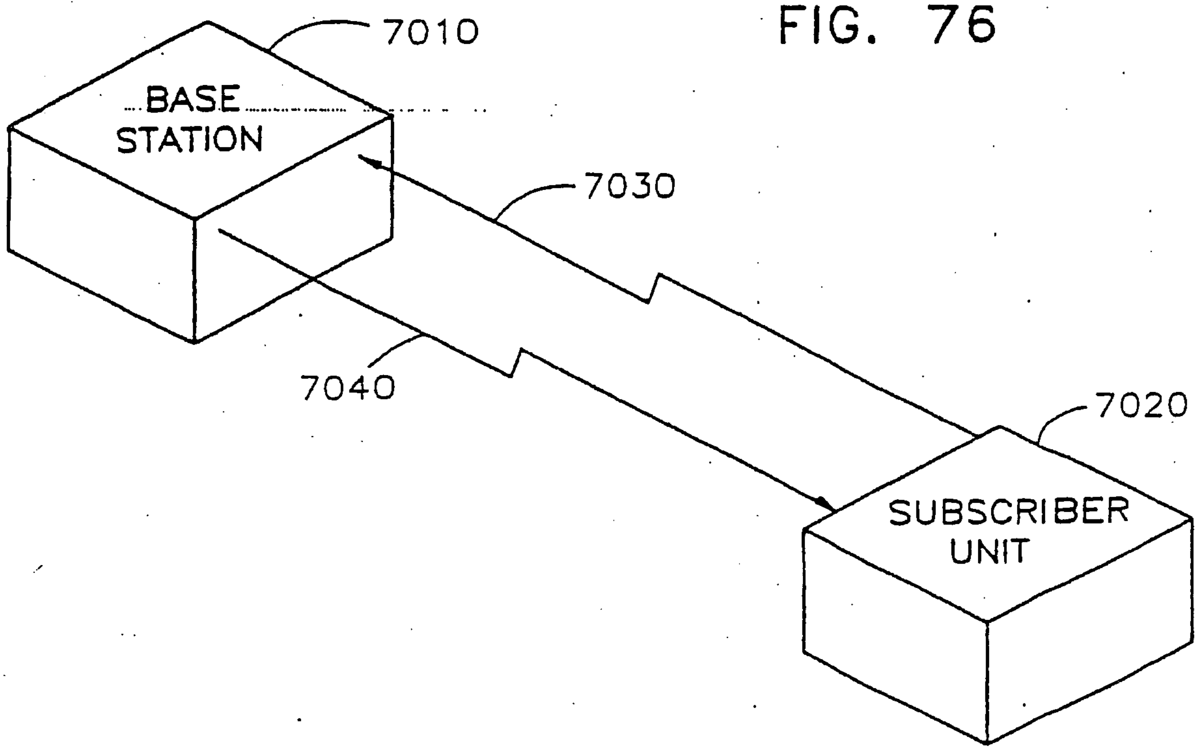

6270
PERFORM OPEN LOOP
6280
CONCURRENTLY PERFORM CLOSED LOOP
6290
ADJUST THE RESULTS OF THE OPEN LOOP BASED ON THE RESULTS OF THE CLOSED LOOP

FIG. 76



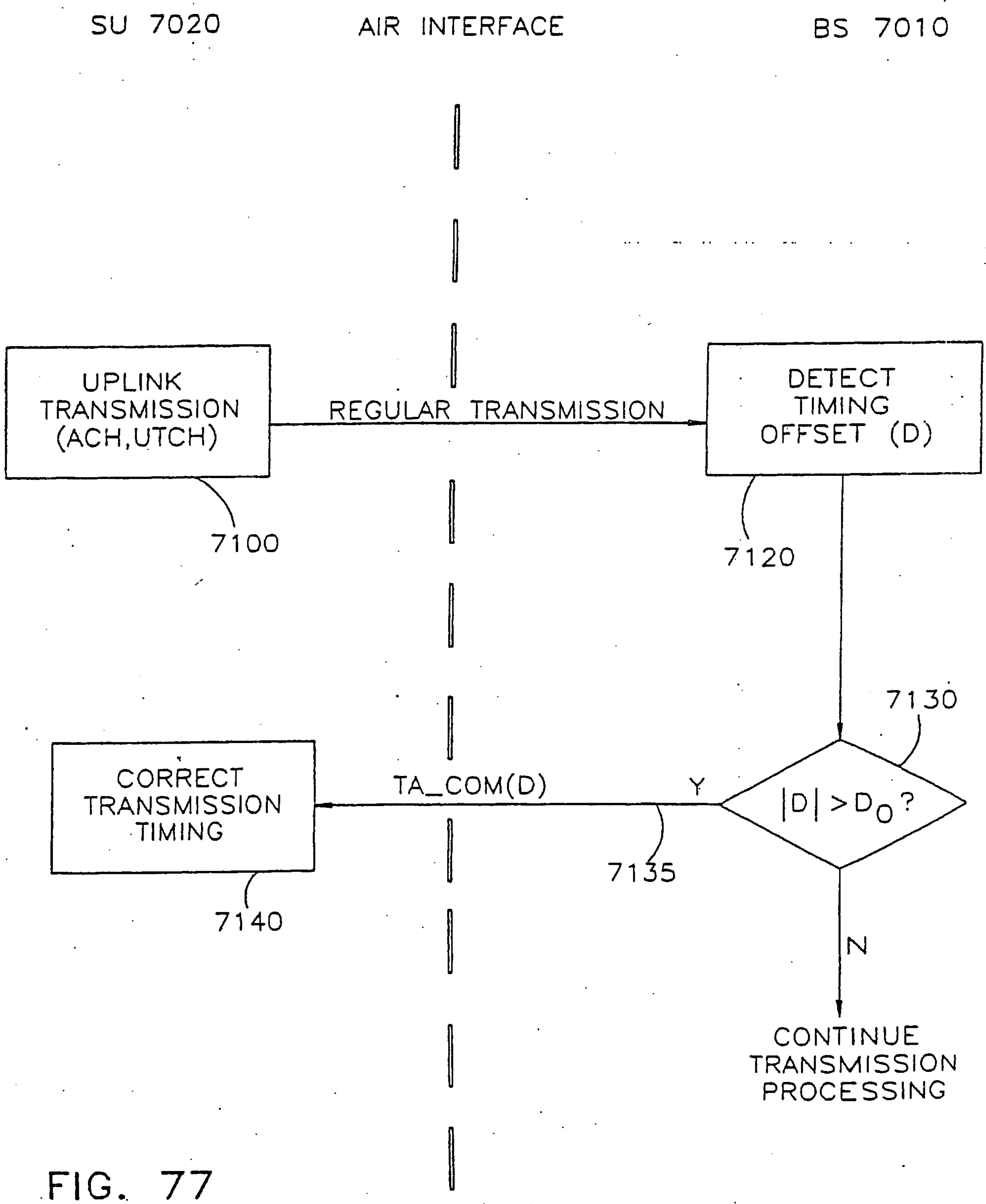
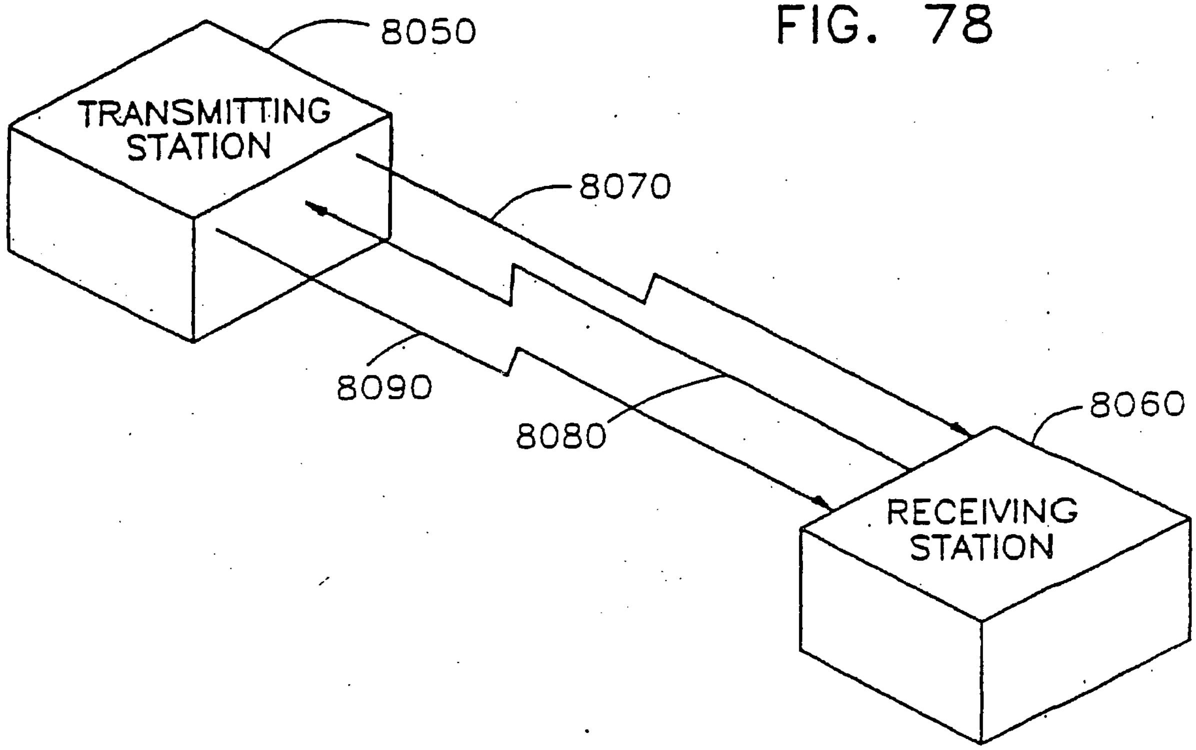

SU 7020
AIR INTERFACE
BS 7010
UPLINK TRANSMISSION (ACH,UTCH)
REGULAR TRANSMISSION
DETECT TIMING OFFSET (D)
7100
7120
7130
CORRECT TRANSMISSION TIMING
TA_COM(D)
Y
|D| > D0 ?
7135
7140
N
CONTINUE TRANSMISSION PROCESSING

FIG. 77

FIG. 78


TRANSMITTING STATION
8050
8070
RECEIVING STATION
8060
8090
8080

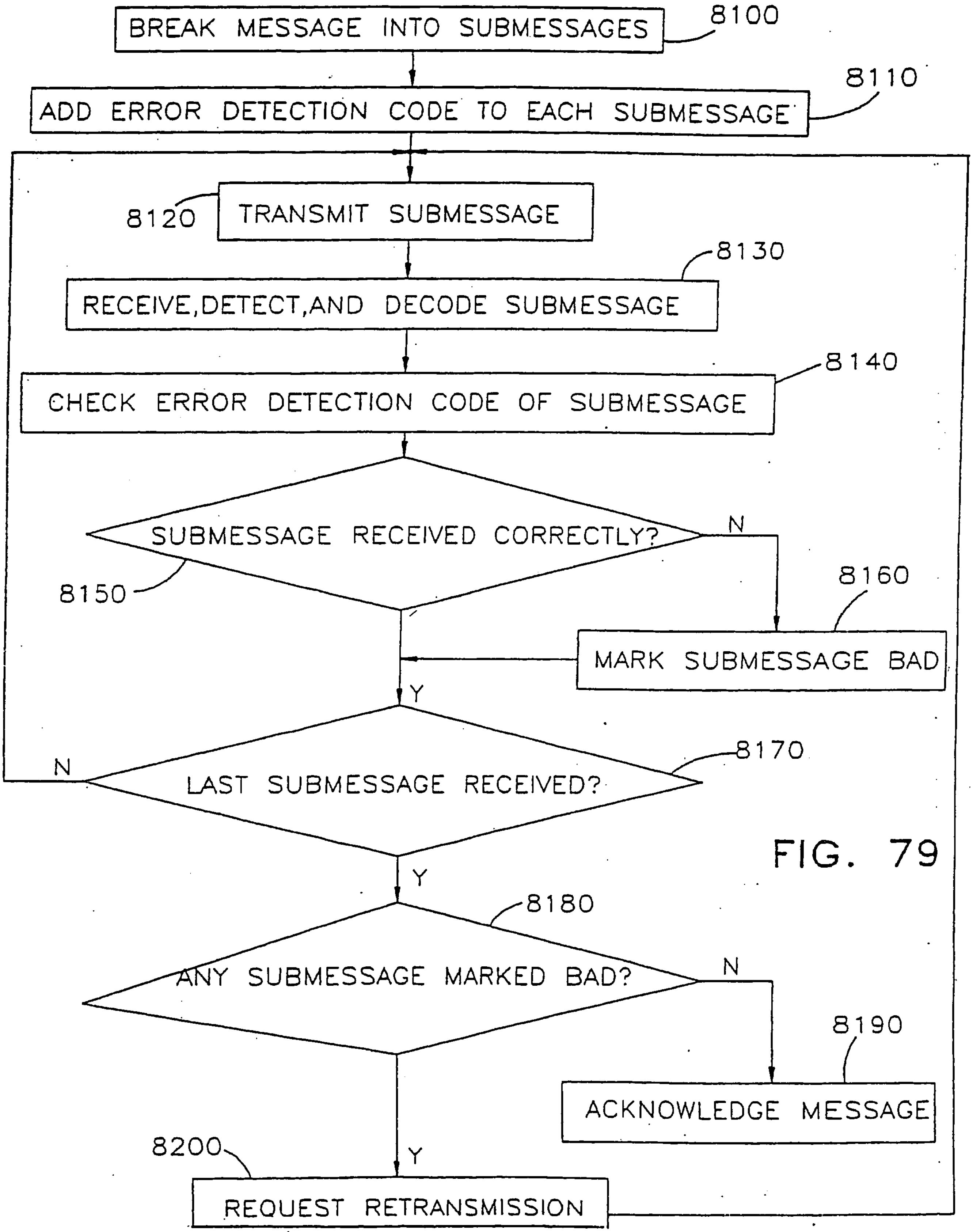

BREAK MESSAGE INTO SUBMESSAGES
8100
ADD ERROR DETECTION CODE TO EACH SUBMESSAGE
8110
TRANSMIT SUBMESSAGE
8120
RECEIVE,DETECT,AND DECODE SUBMESSAGE
8130
CHECK ERROR DETECTION CODE OF SUBMESSAGE
8140
SUBMESSAGE RECEIVED CORRECTLY?
8150
N
MARK SUBMESSAGE BAD
8160
Y
LAST SUBMESSAGE RECEIVED?
8170
N
Y
ANY SUBMESSAGE MARKED BAD?
8180
N
ACKNOWLEDGE MESSAGE
8190
Y
REQUEST RETRANSMISSION
8200

FIG. 79

FIG. 80

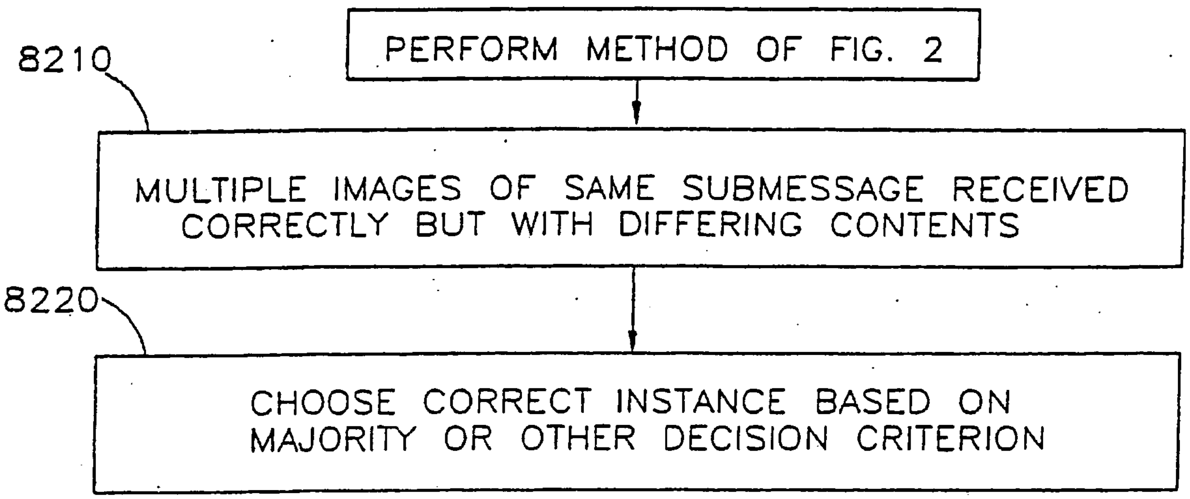
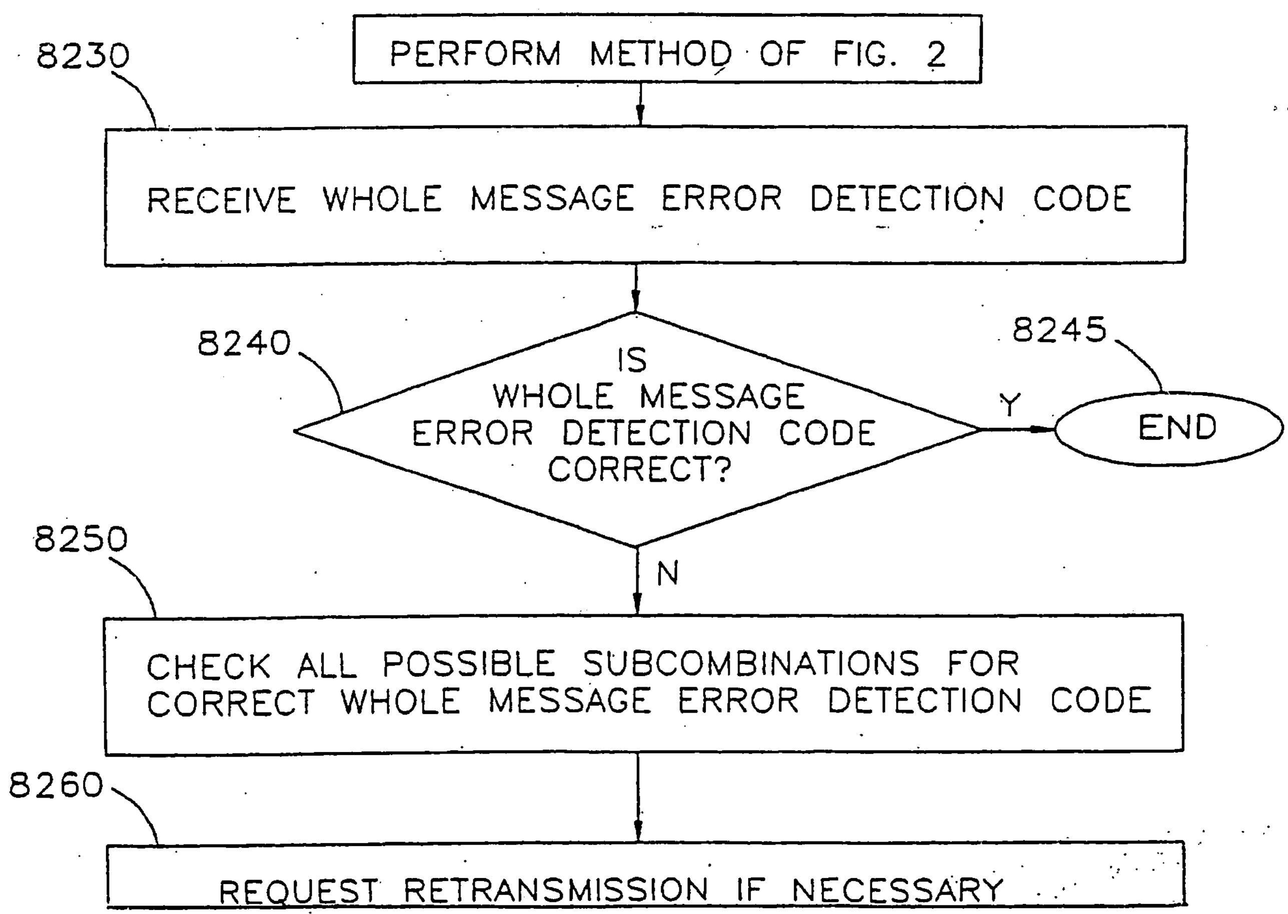

PERFORM METHOD OF FIG. 2
8210
MULTIPLE IMAGES OF SAME SUBMESSAGE RECEIVED CORRECTLY BUT WITH DIFFERING CONTENTS
8220
CHOOSE CORRECT INSTANCE BASED ON MAJORITY OR OTHER DECISION CRITERION

FIG. 81


PERFORM METHOD OF FIG. 2
8230
RECEIVE WHOLE MESSAGE ERROR DETECTION CODE
8240
IS WHOLE MESSAGE ERROR DETECTION CODE CORRECT?
Y
8245
END
N
8250
CHECK ALL POSSIBLE SUBCOMBINATIONS FOR CORRECT WHOLE MESSAGE ERROR DETECTION CODE
8260
REQUEST RETRANSMISSION IF NECESSARY

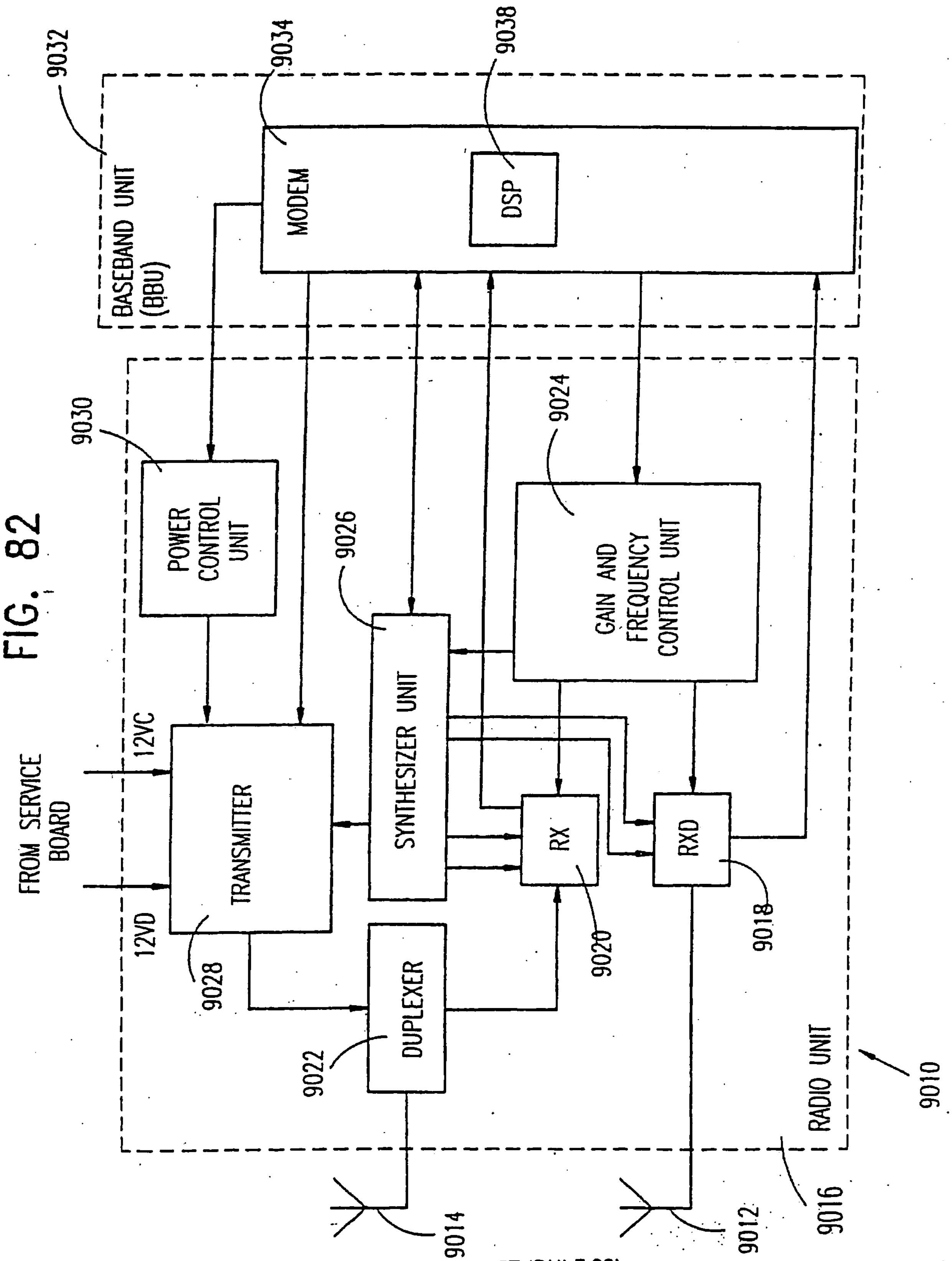
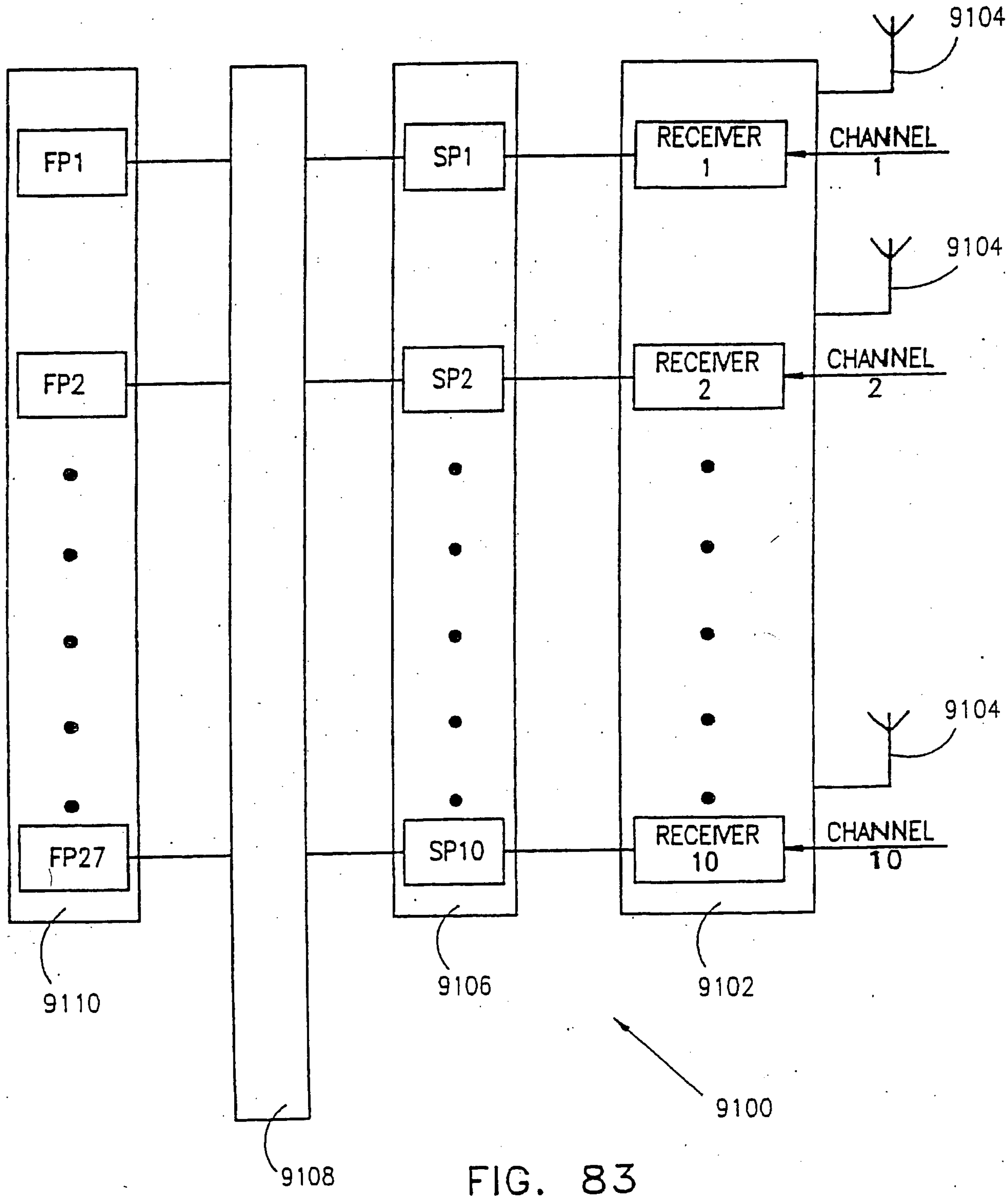

9032
BASEBAND UNIT (BBU)
9034
MODEM
9038
DSP
9030
POWER CONTROL UNIT
9024
GAIN AND FREQUENCY CONTROL UNIT
9026
SYNTHESIZER UNIT
FROM SERVICE BOARD
12VC
12VD
TRANSMITTER
9028
RX
9020
RXD
9018
DUPLEXER
9022
RADIO UNIT
9010
9016
9014
9012

FIG. 82



9104
9104
9104
FP1
FP2
FP27
SP1
SP2
SP10
RECEIVER 1
RECEIVER 2
RECEIVER 10
CHANNEL 1
CHANNEL 2
CHANNEL 10
9110
9108
9106
9102
9100

FIG. 83

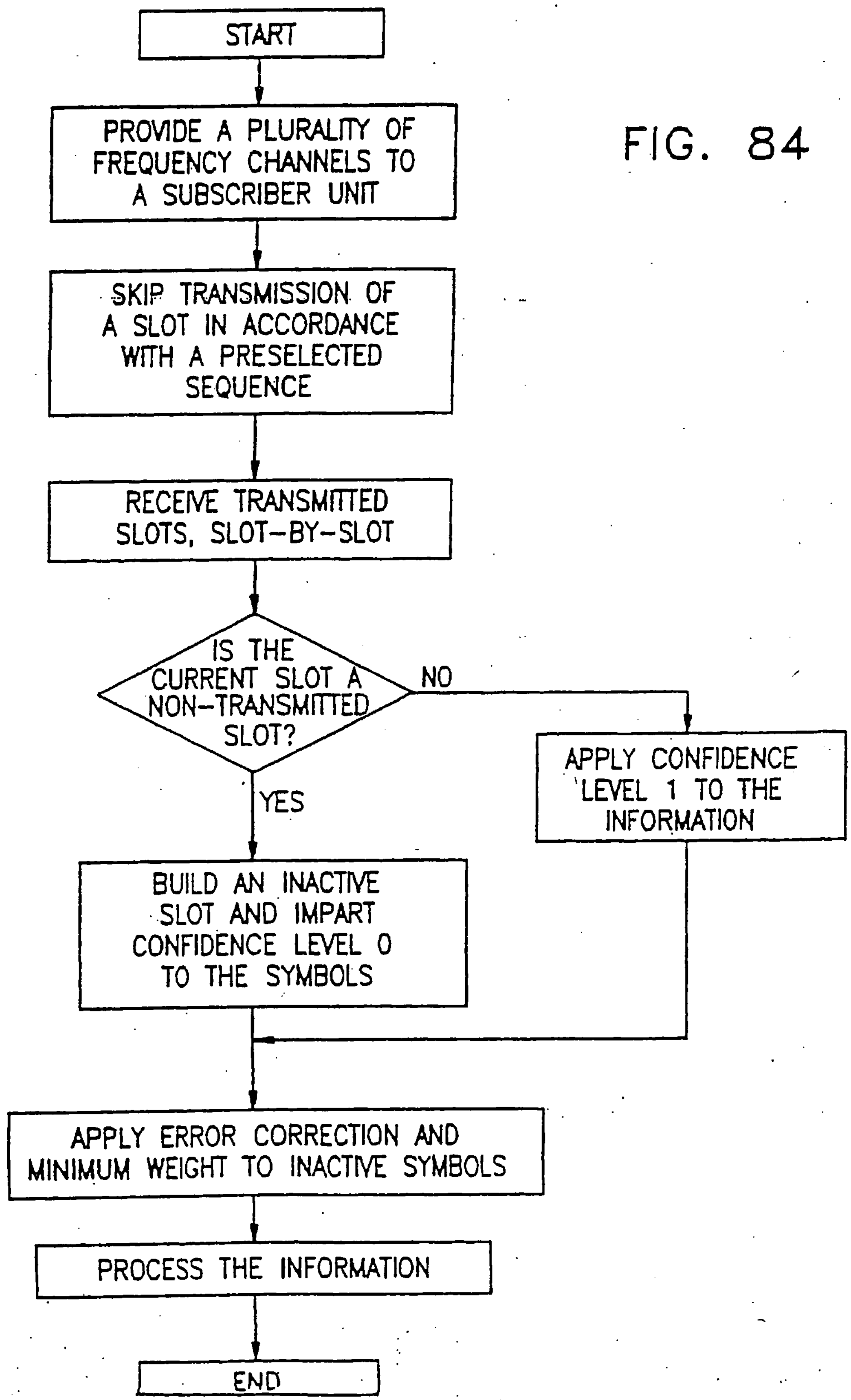

START
PROVIDE A PLURALITY OF FREQUENCY CHANNELS TO A SUBSCRIBER UNIT
SKIP TRANSMISSION OF A SLOT IN ACCORDANCE WITH A PRESELECTED SEQUENCE
RECEIVE TRANSMITTED SLOTS, SLOT-BY-SLOT
IS THE CURRENT SLOT A NON-TRANSMITTED SLOT?
NO
YES
APPLY CONFIDENCE LEVEL 1 TO THE INFORMATION
BUILD AN INACTIVE SLOT AND IMPART CONFIDENCE LEVEL 0 TO THE SYMBOLS
APPLY ERROR CORRECTION AND MINIMUM WEIGHT TO INACTIVE SYMBOLS
PROCESS THE INFORMATION
END

FIG. 84

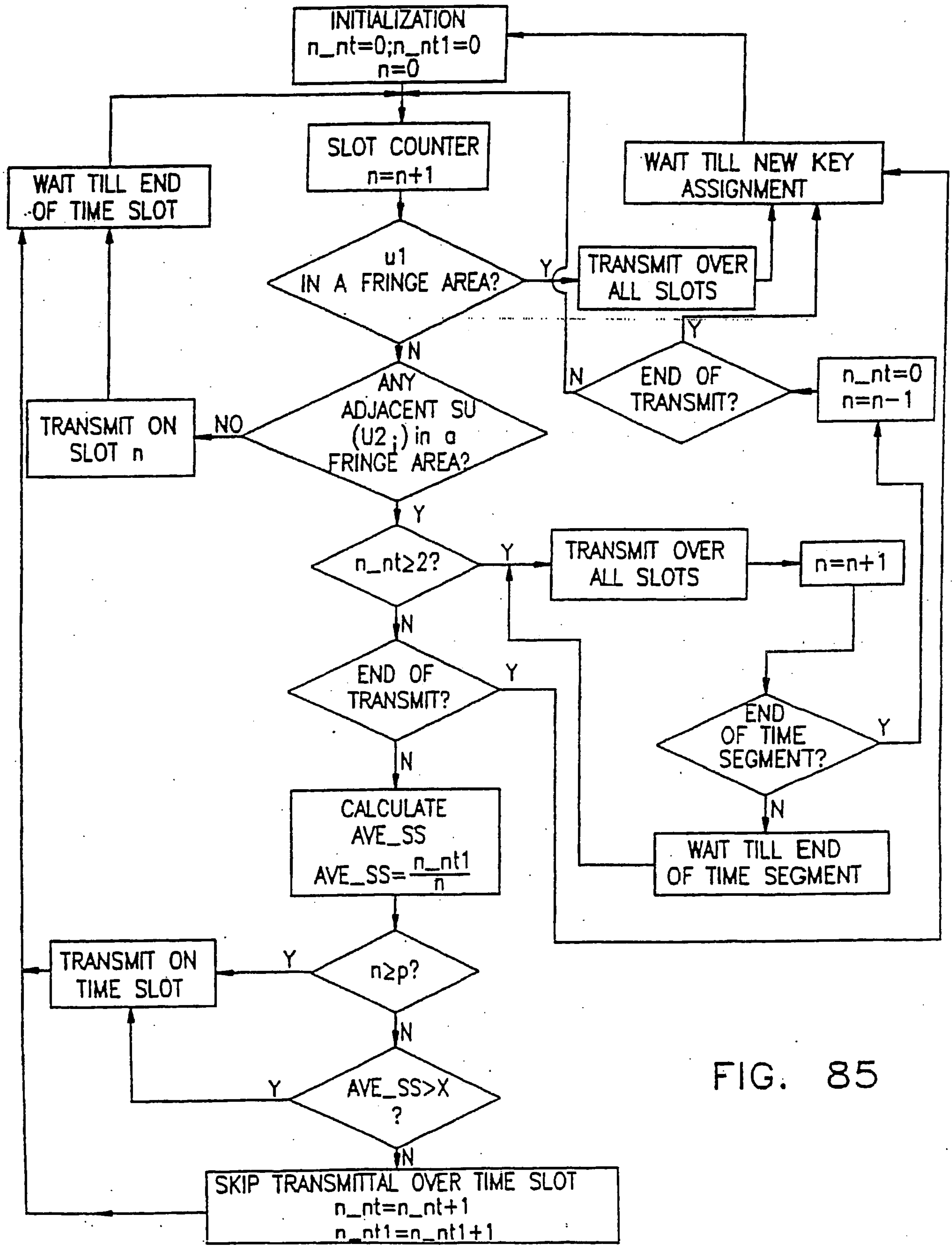

INITIALIZATION
n_nt=0;n_nt1=0
n=0
SLOT COUNTER
n=n+1
WAIT TILL END OF TIME SLOT
u1 IN A FRINGE AREA?
Y
N
TRANSMIT OVER ALL SLOTS
WAIT TILL NEW KEY ASSIGNMENT
END OF TRANSMIT?
n_nt=0
n=n-1
ANY ADJACENT SU (U2 i) in a FRINGE AREA?
NO
TRANSMIT ON SLOT n
n_nt≥2?
TRANSMIT OVER ALL SLOTS
n=n+1
END OF TRANSMIT?
END OF TIME SEGMENT?
WAIT TILL END OF TIME SEGMENT
CALCULATE AVE_SS
AVE_SS=n_nt1/n
n≥p?
TRANSMIT ON TIME SLOT
AVE_SS>X ?
SKIP TRANSMITTAL OVER TIME SLOT
n_nt=n_nt+1
n_nt1=n_nt1+1

FIG. 85

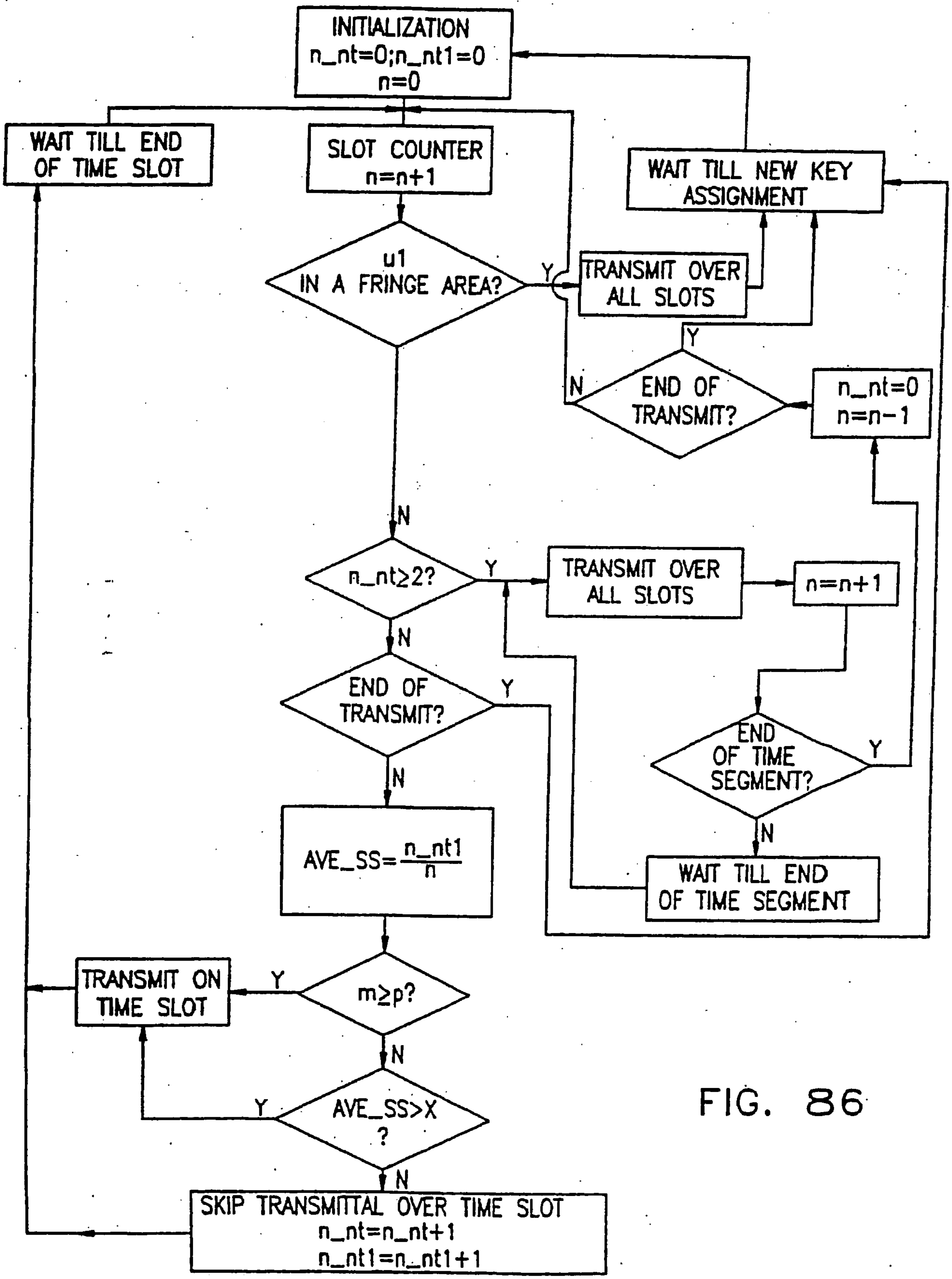

INITIALIZATION
n_nt=0;n_nt1=0
n=0
WAIT TILL END OF TIME SLOT
SLOT COUNTER n=n+1
WAIT TILL NEW KEY ASSIGNMENT
u1 IN A FRINGE AREA?
Y
TRANSMIT OVER ALL SLOTS
Y
N
END OF TRANSMIT?
n_nt=0
n=n-1
N
n_nt≥2?
Y
TRANSMIT OVER ALL SLOTS
n=n+1
N
END OF TRANSMIT?
Y
END OF TIME SEGMENT?
Y
N
N
AVE_SS=n_nt1/n
WAIT TILL END OF TIME SEGMENT
TRANSMIT ON TIME SLOT
Y
m≥p?
N
Y
AVE_SS>X ?
N
SKIP TRANSMITTAL OVER TIME SLOT
n_nt=n_nt+1
n_nt1=n_nt1+1

FIG. 86

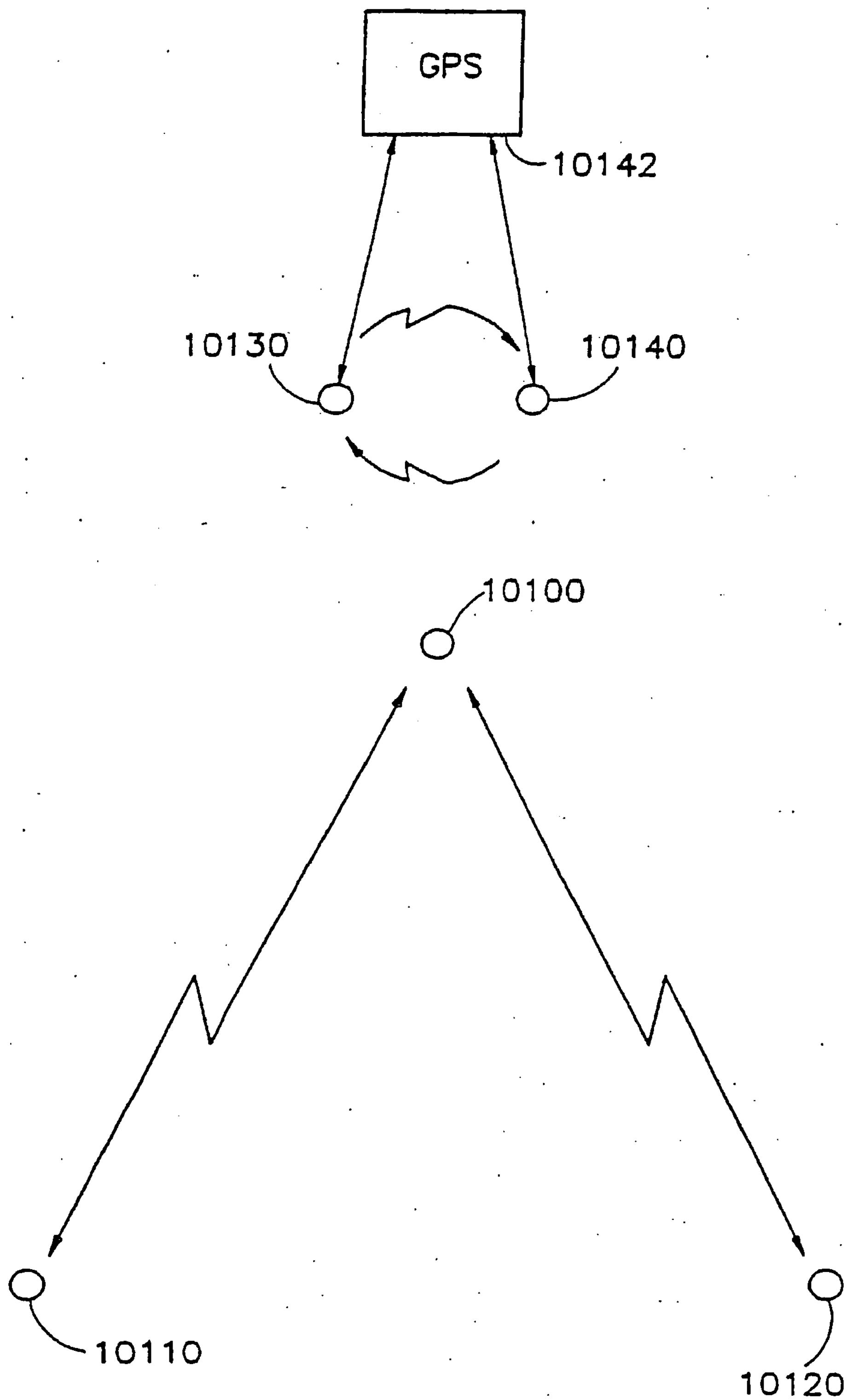
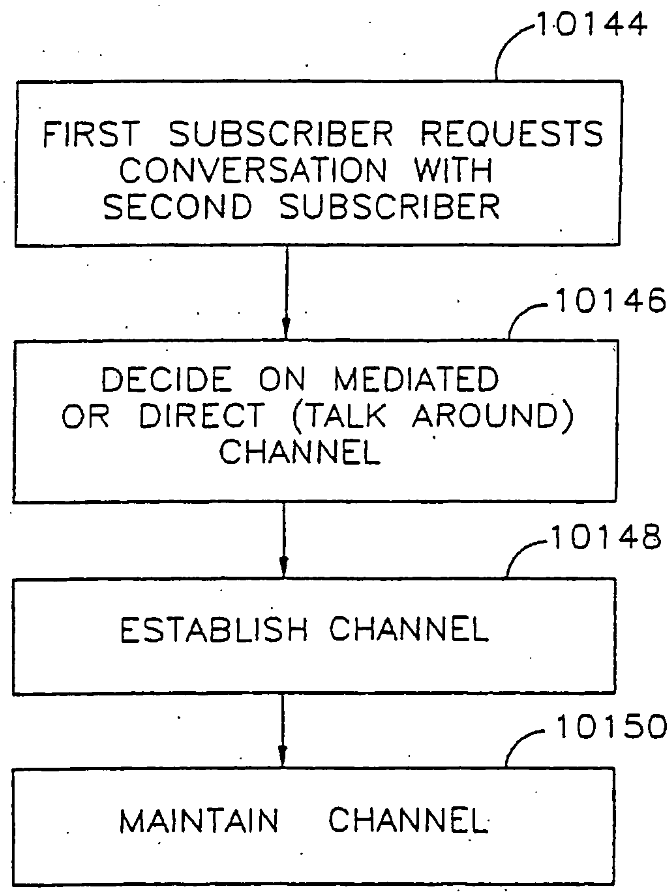


FIG. 87



10144
FIRST SUBSCRIBER REQUESTS CONVERSATION WITH SECOND SUBSCRIBER
10146
DECIDE ON MEDIATED OR DIRECT (TALK AROUND) CHANNEL
10148
ESTABLISH CHANNEL
10150
MAINTAIN CHANNEL

FIG. 88A

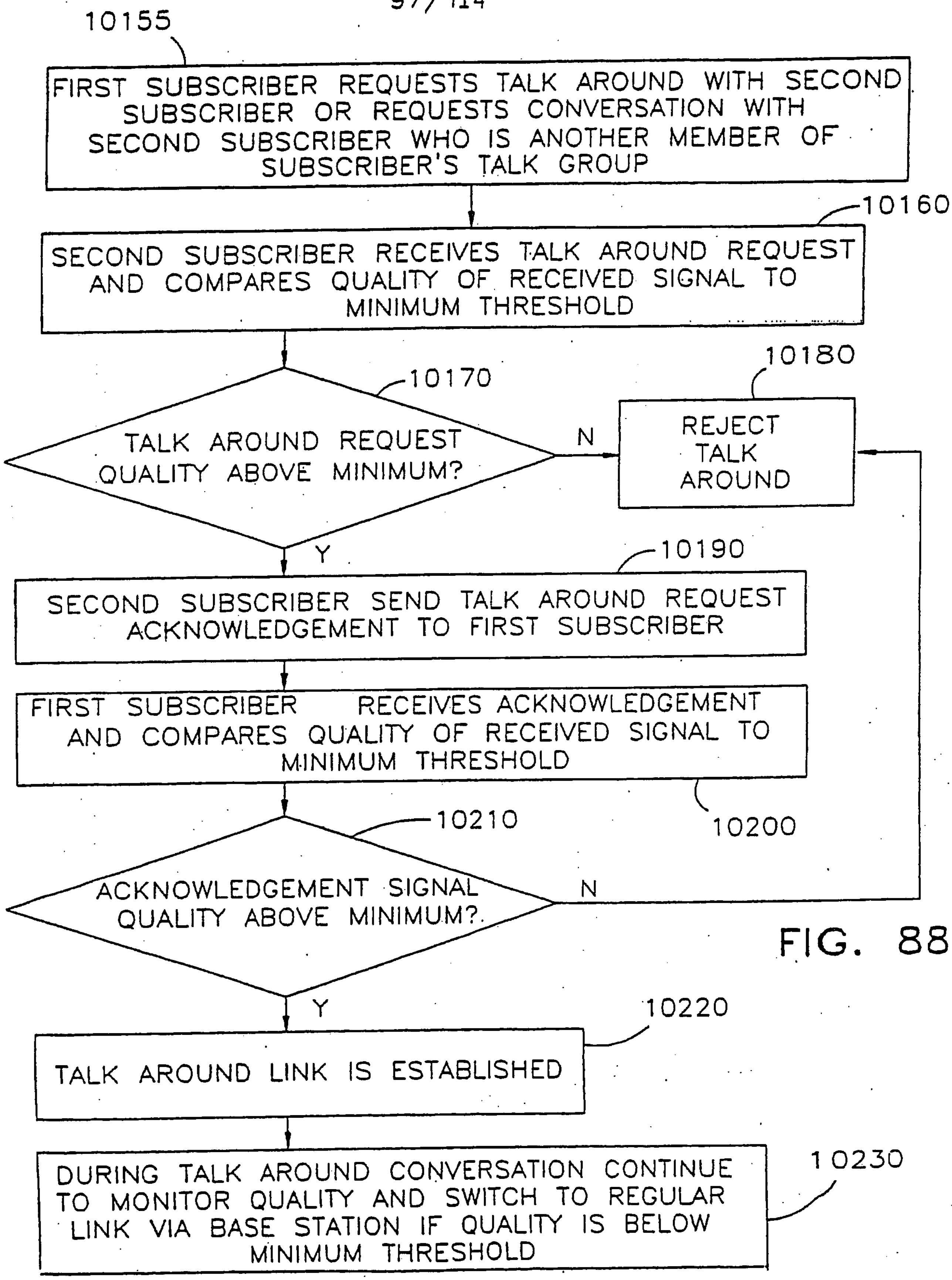
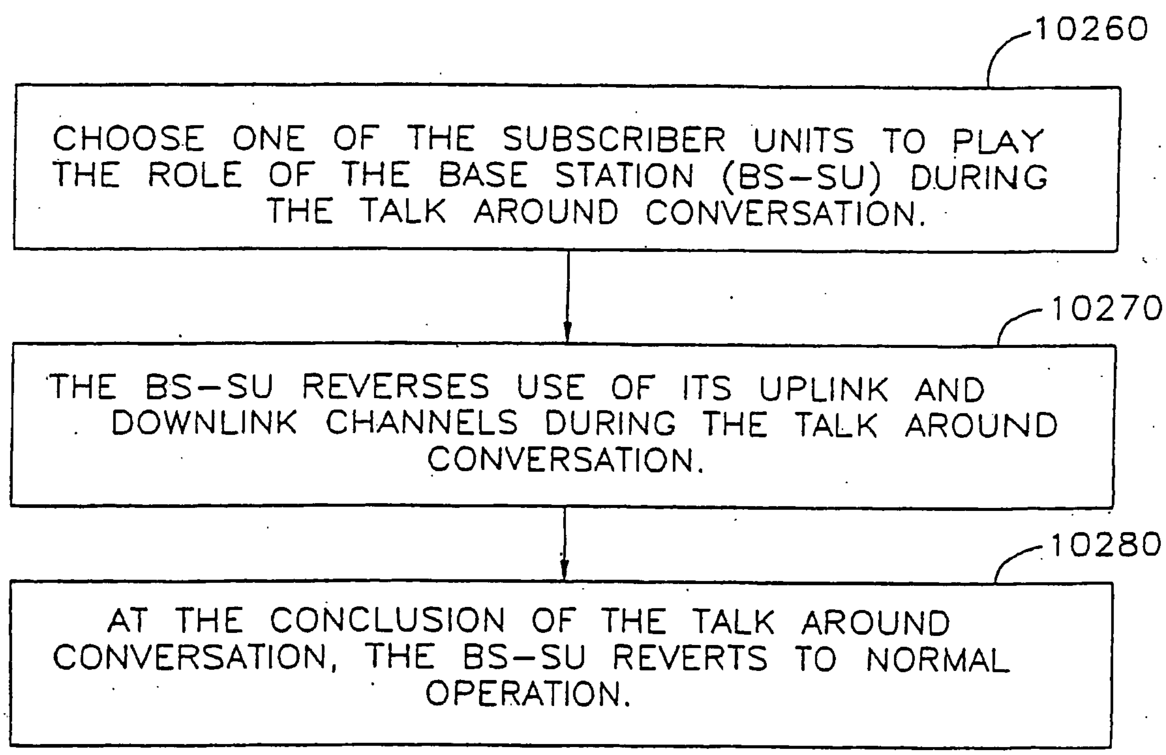

10155
FIRST SUBSCRIBER REQUESTS TALK AROUND WITH SECOND SUBSCRIBER OR REQUESTS CONVERSATION WITH SECOND SUBSCRIBER WHO IS ANOTHER MEMBER OF SUBSCRIBER'S TALK GROUP
10160
SECOND SUBSCRIBER RECEIVES TALK AROUND REQUEST AND COMPARES QUALITY OF RECEIVED SIGNAL TO MINIMUM THRESHOLD
10170
TALK AROUND REQUEST QUALITY ABOVE MINIMUM?
N
10180
REJECT TALK AROUND
Y
10190
SECOND SUBSCRIBER SEND TALK AROUND REQUEST ACKNOWLEDGEMENT TO FIRST SUBSCRIBER
FIRST SUBSCRIBER RECEIVES ACKNOWLEDGEMENT AND COMPARES QUALITY OF RECEIVED SIGNAL TO MINIMUM THRESHOLD
10200
10210
ACKNOWLEDGEMENT SIGNAL QUALITY ABOVE MINIMUM?
N
Y
10220
TALK AROUND LINK IS ESTABLISHED
10230
DURING TALK AROUND CONVERSATION CONTINUE TO MONITOR QUALITY AND SWITCH TO REGULAR LINK VIA BASE STATION IF QUALITY IS BELOW MINIMUM THRESHOLD

FIG. 88B

FIG. 89


10260
CHOOSE ONE OF THE SUBSCRIBER UNITS TO PLAY THE ROLE OF THE BASE STATION (BS-SU) DURING THE TALK AROUND CONVERSATION.
10270
THE BS-SU REVERSES USE OF ITS UPLINK AND DOWNLINK CHANNELS DURING THE TALK AROUND CONVERSATION.
10280
AT THE CONCLUSION OF THE TALK AROUND CONVERSATION, THE BS-SU REVERTS TO NORMAL OPERATION.

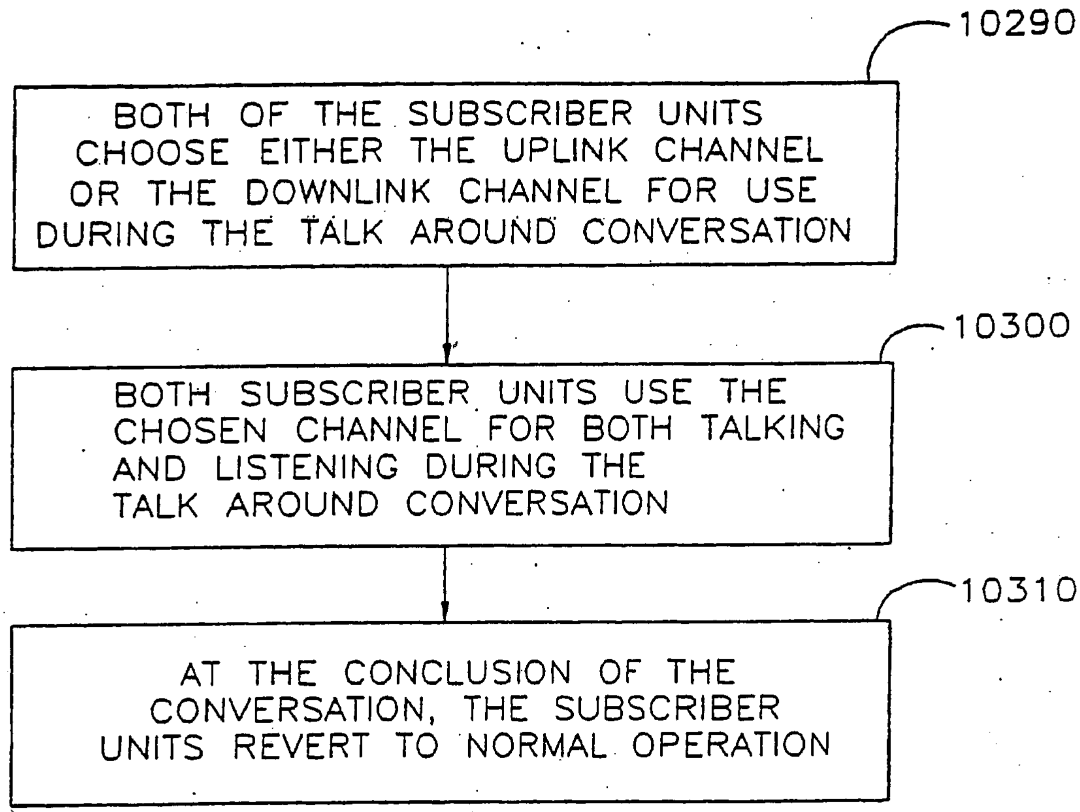

10290
BOTH OF THE SUBSCRIBER UNITS CHOOSE EITHER THE UPLINK CHANNEL OR THE DOWNLINK CHANNEL FOR USE DURING THE TALK AROUND CONVERSATION
10300
BOTH SUBSCRIBER UNITS USE THE CHOSEN CHANNEL FOR BOTH TALKING AND LISTENING DURING THE TALK AROUND CONVERSATION
10310
AT THE CONCLUSION OF THE CONVERSATION, THE SUBSCRIBER UNITS REVERT TO NORMAL OPERATION

FIG. 90

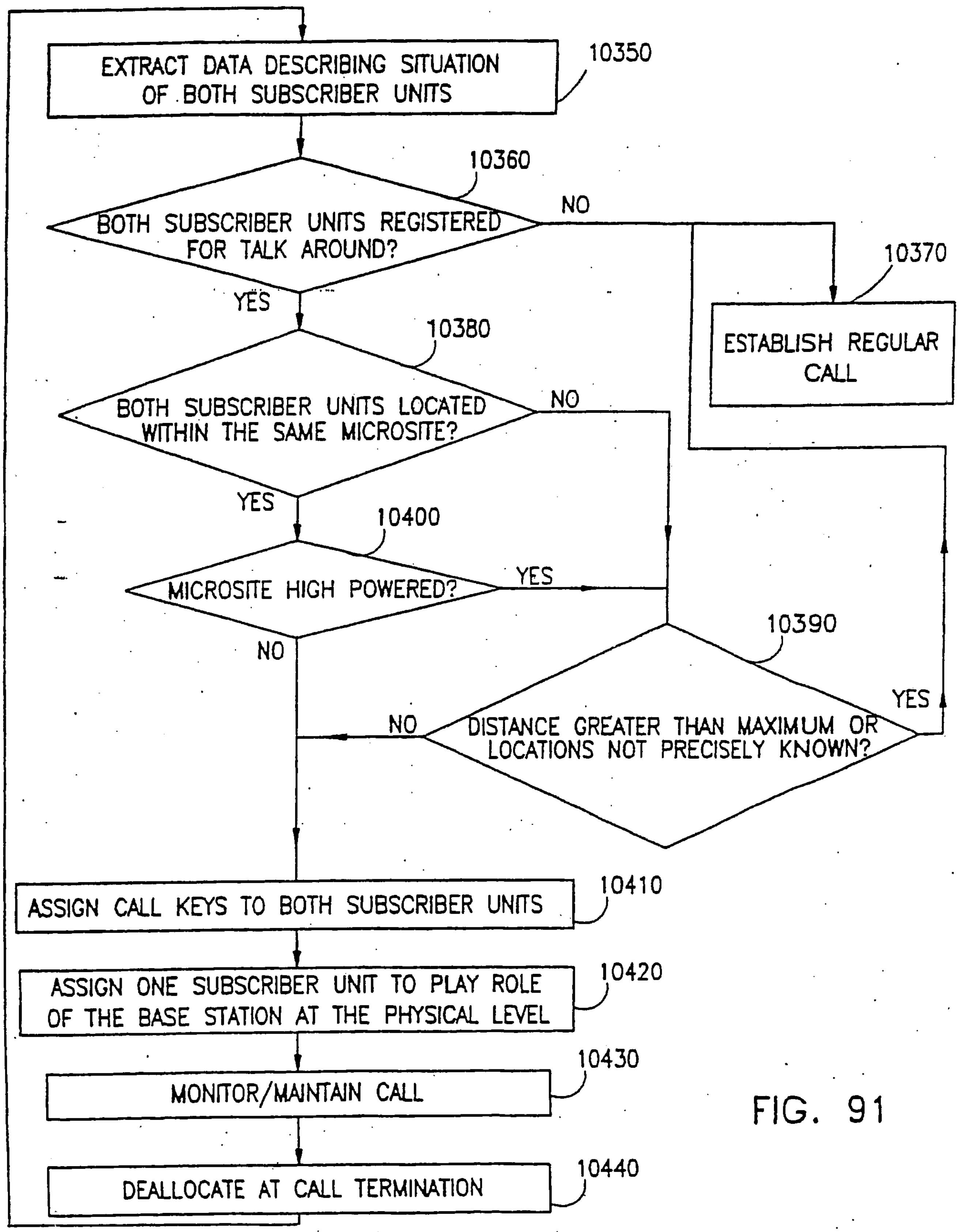

EXTRACT DATA DESCRIBING SITUATION OF BOTH SUBSCRIBER UNITS
10350
10360
BOTH SUBSCRIBER UNITS REGISTERED FOR TALK AROUND?
NO
YES
10370
ESTABLISH REGULAR CALL
10380
BOTH SUBSCRIBER UNITS LOCATED WITHIN THE SAME MICROSITE?
NO
YES
10400
MICROSITE HIGH POWERED?
YES
NO
10390
DISTANCE GREATER THAN MAXIMUM OR LOCATIONS NOT PRECISELY KNOWN?
YES
NO
ASSIGN CALL KEYS TO BOTH SUBSCRIBER UNITS
10410
ASSIGN ONE SUBSCRIBER UNIT TO PLAY ROLE OF THE BASE STATION AT THE PHYSICAL LEVEL
10420
MONITOR/MAINTAIN CALL
10430
DEALLOCATE AT CALL TERMINATION
10440

FIG. 91

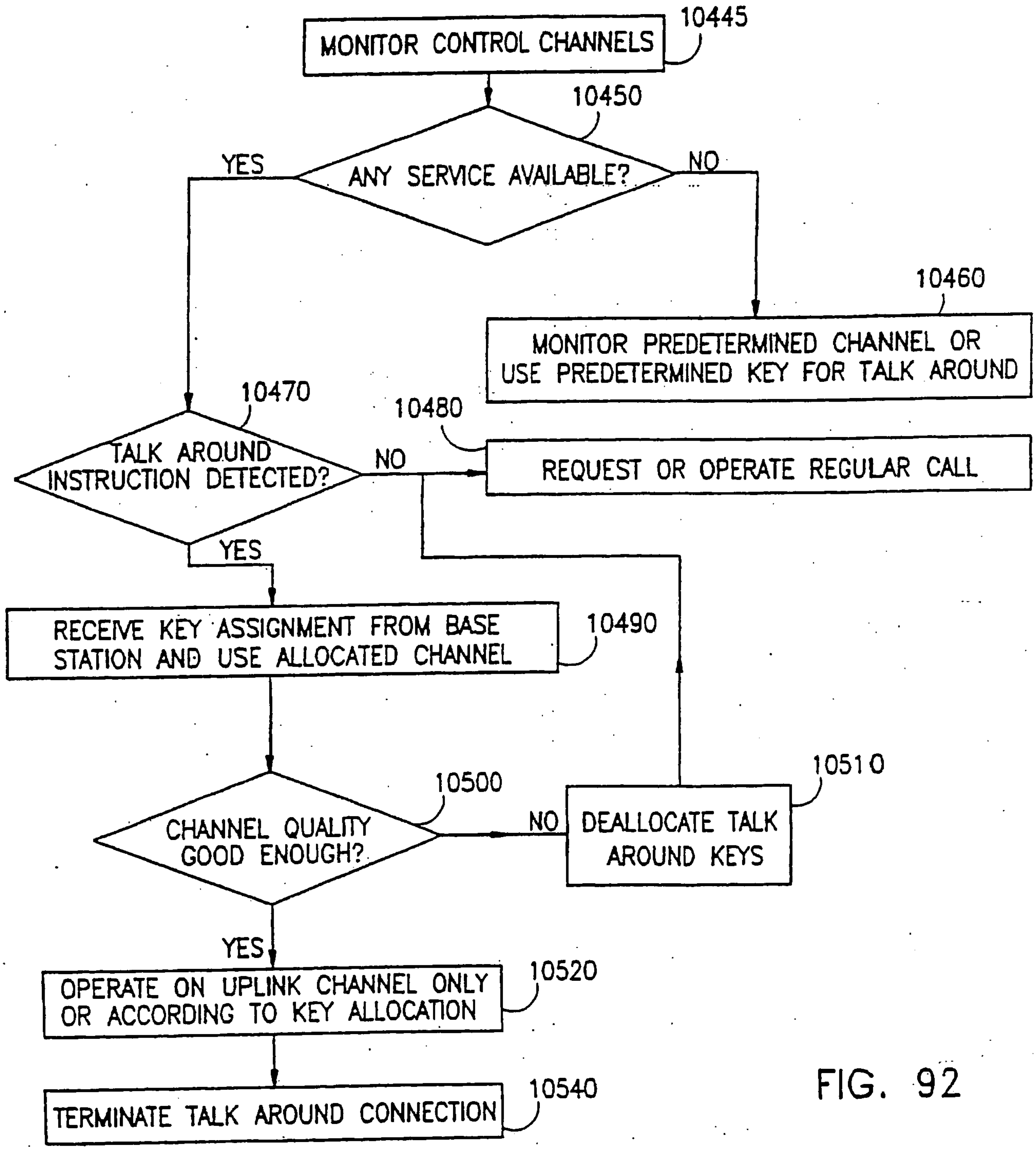

MONITOR CONTROL CHANNELS
10445
10450
ANY SERVICE AVAILABLE?
YES
NO
10460
MONITOR PREDETERMINED CHANNEL OR USE PREDETERMINED KEY FOR TALK AROUND
10470
TALK AROUND INSTRUCTION DETECTED?
NO
10480
REQUEST OR OPERATE REGULAR CALL
YES
RECEIVE KEY ASSIGNMENT FROM BASE STATION AND USE ALLOCATED CHANNEL
10490
10500
CHANNEL QUALITY GOOD ENOUGH?
NO
DEALLOCATE TALK AROUND KEYS
10510
YES
OPERATE ON UPLINK CHANNEL ONLY OR ACCORDING TO KEY ALLOCATION
10520
TERMINATE TALK AROUND CONNECTION
10540

FIG. 92

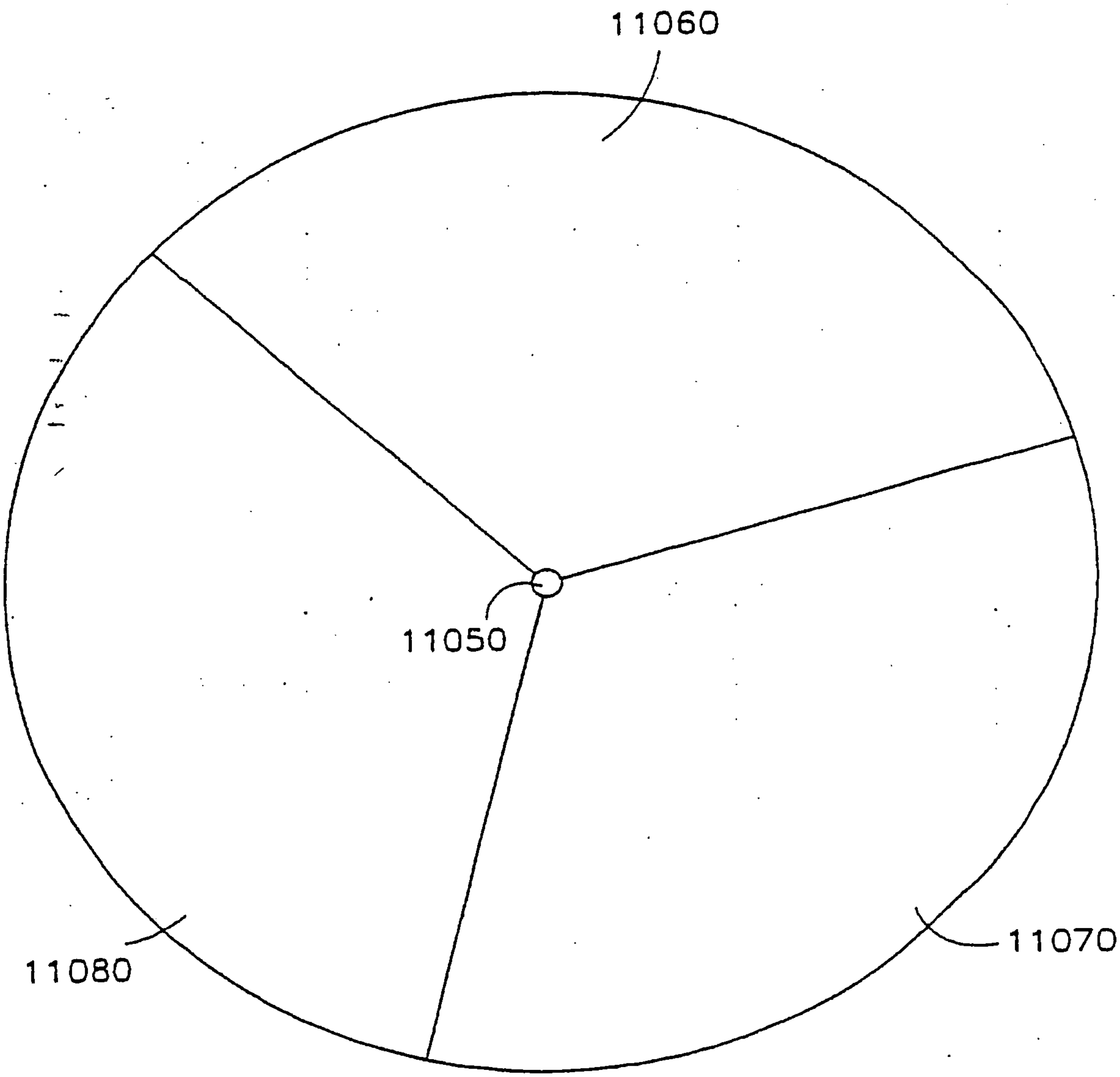


FIG. 93
PRIOR ART

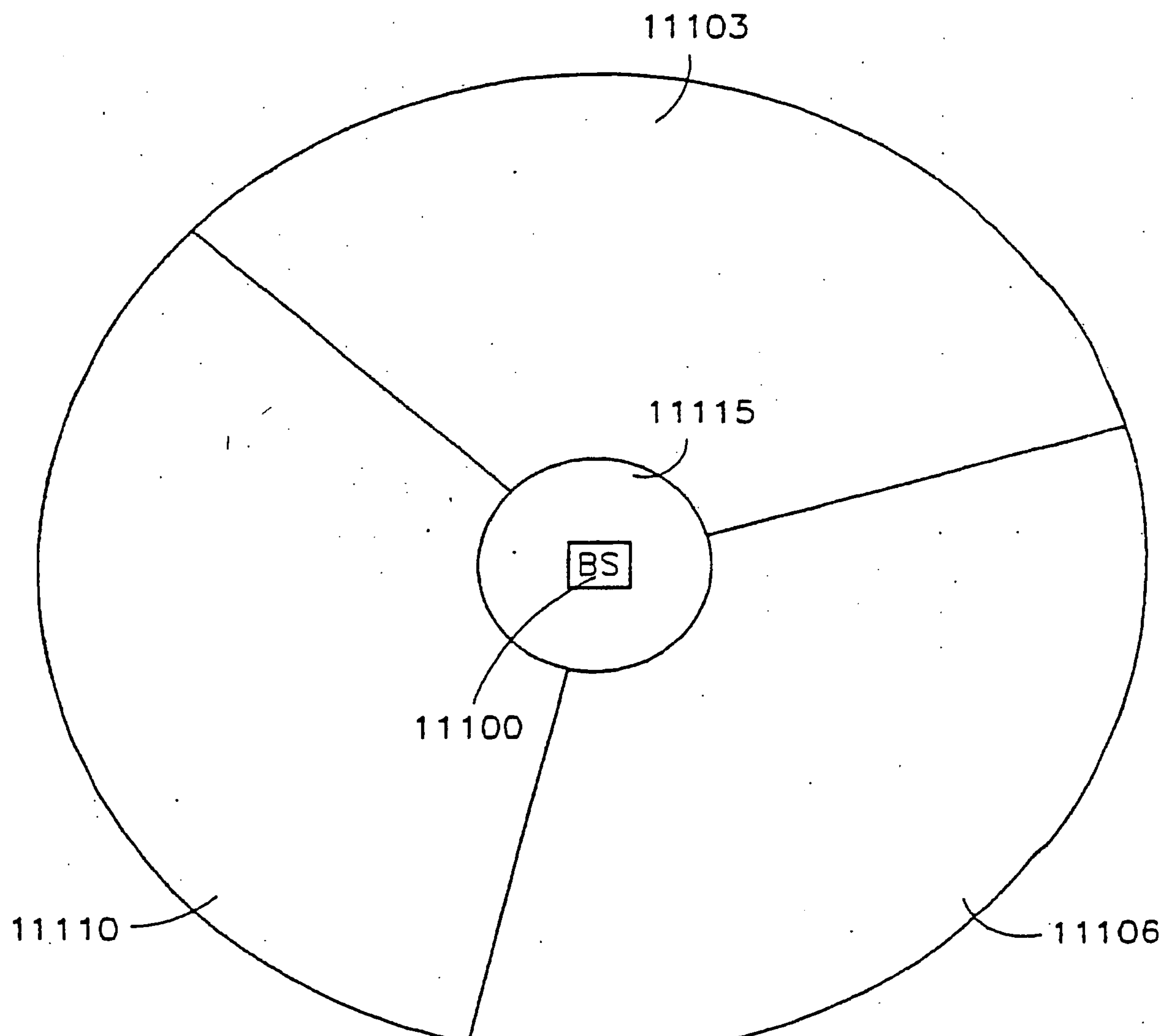


FIG. 94

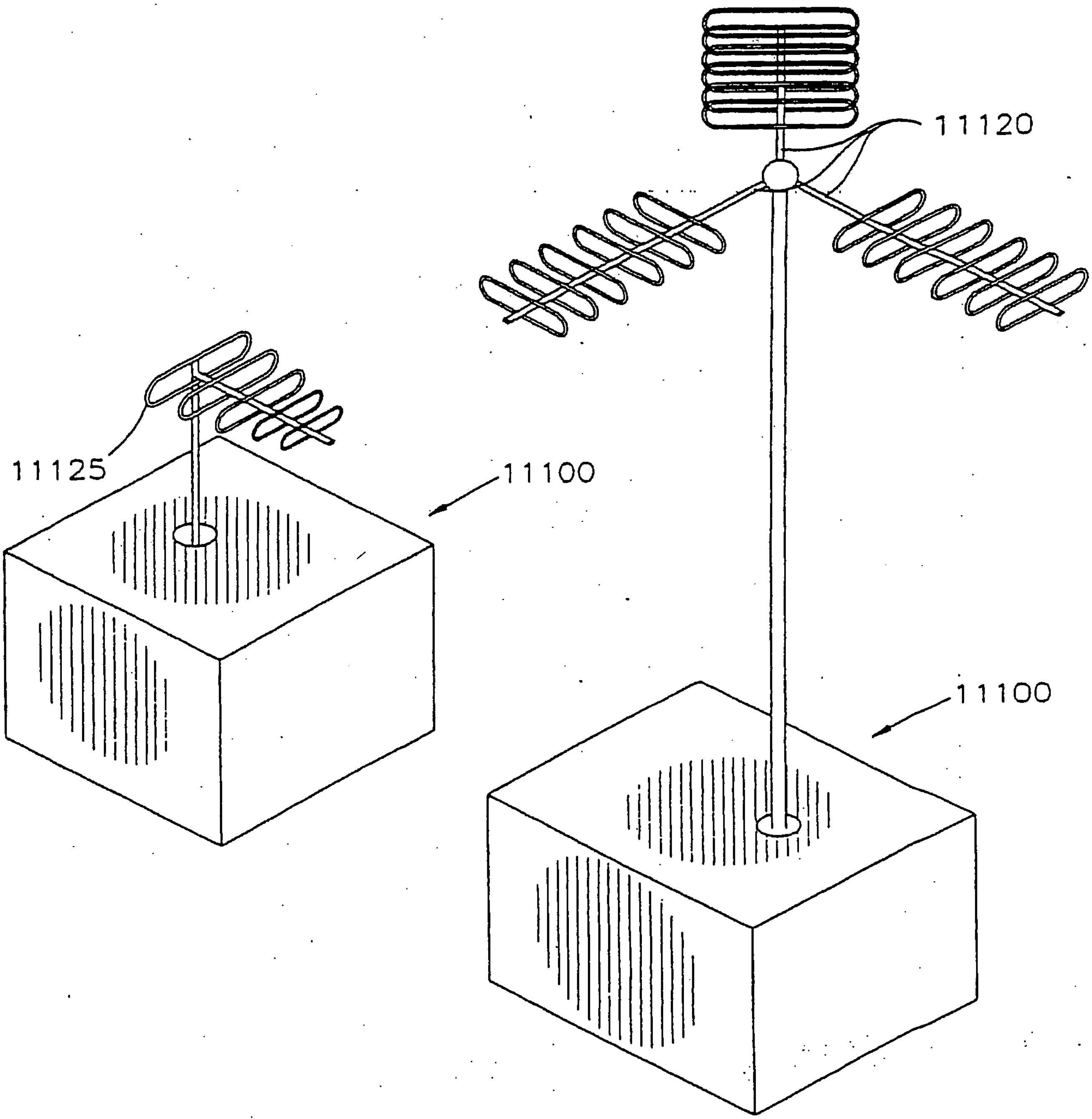


FIG. 95A

FIG. 95B

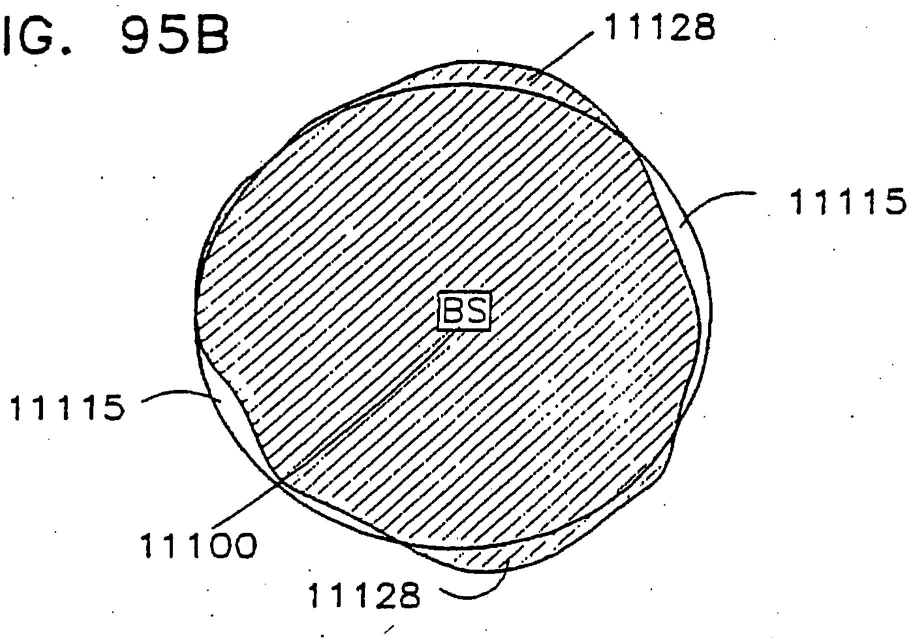
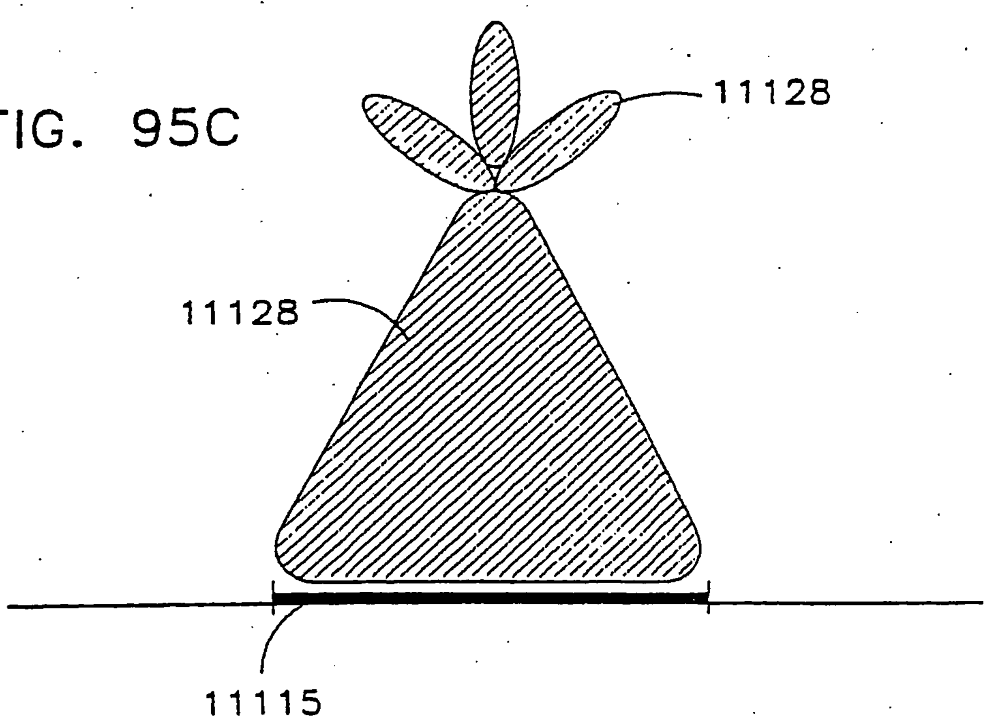


FIG. 95C



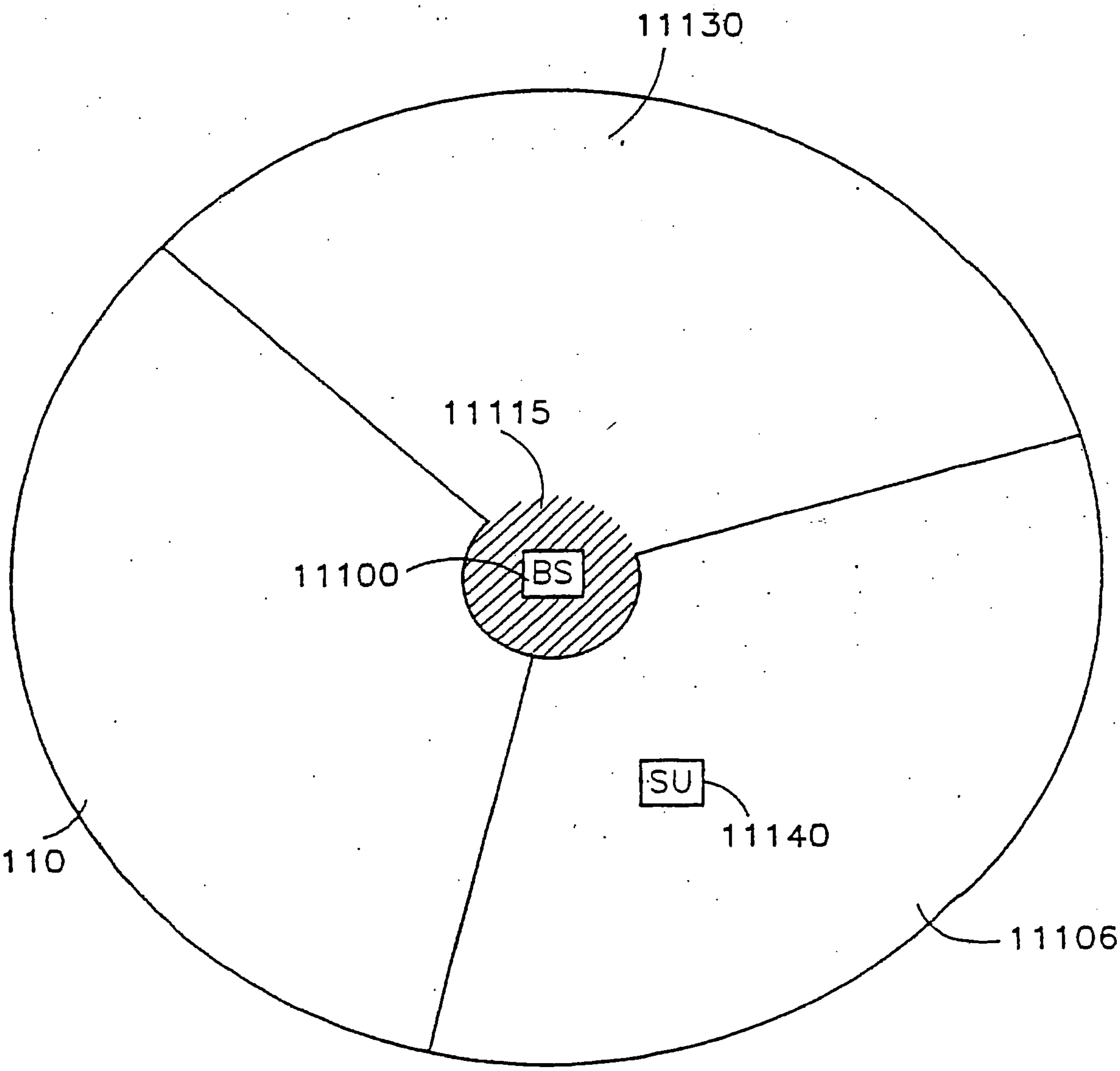


FIG. 96

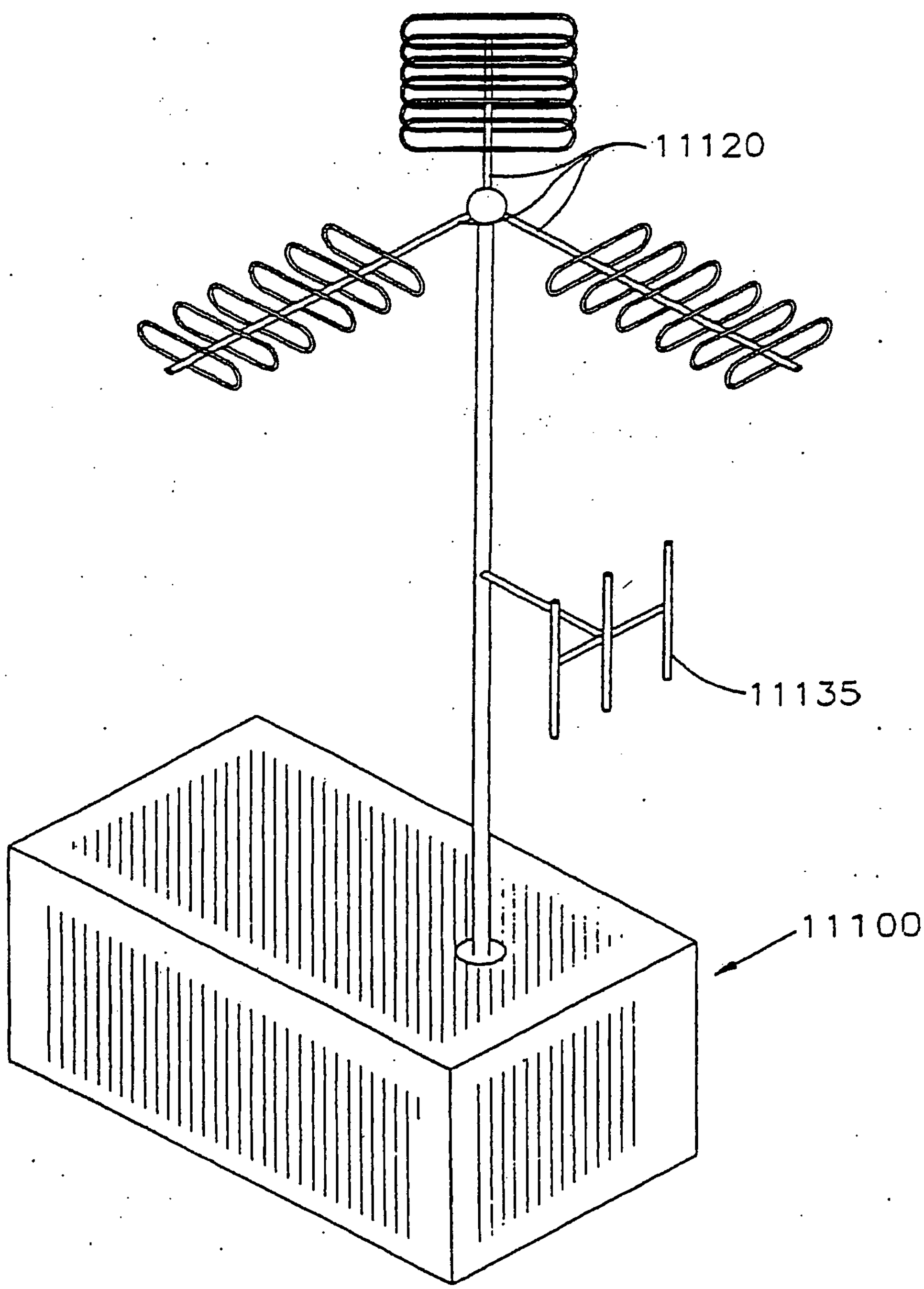


FIG. 97

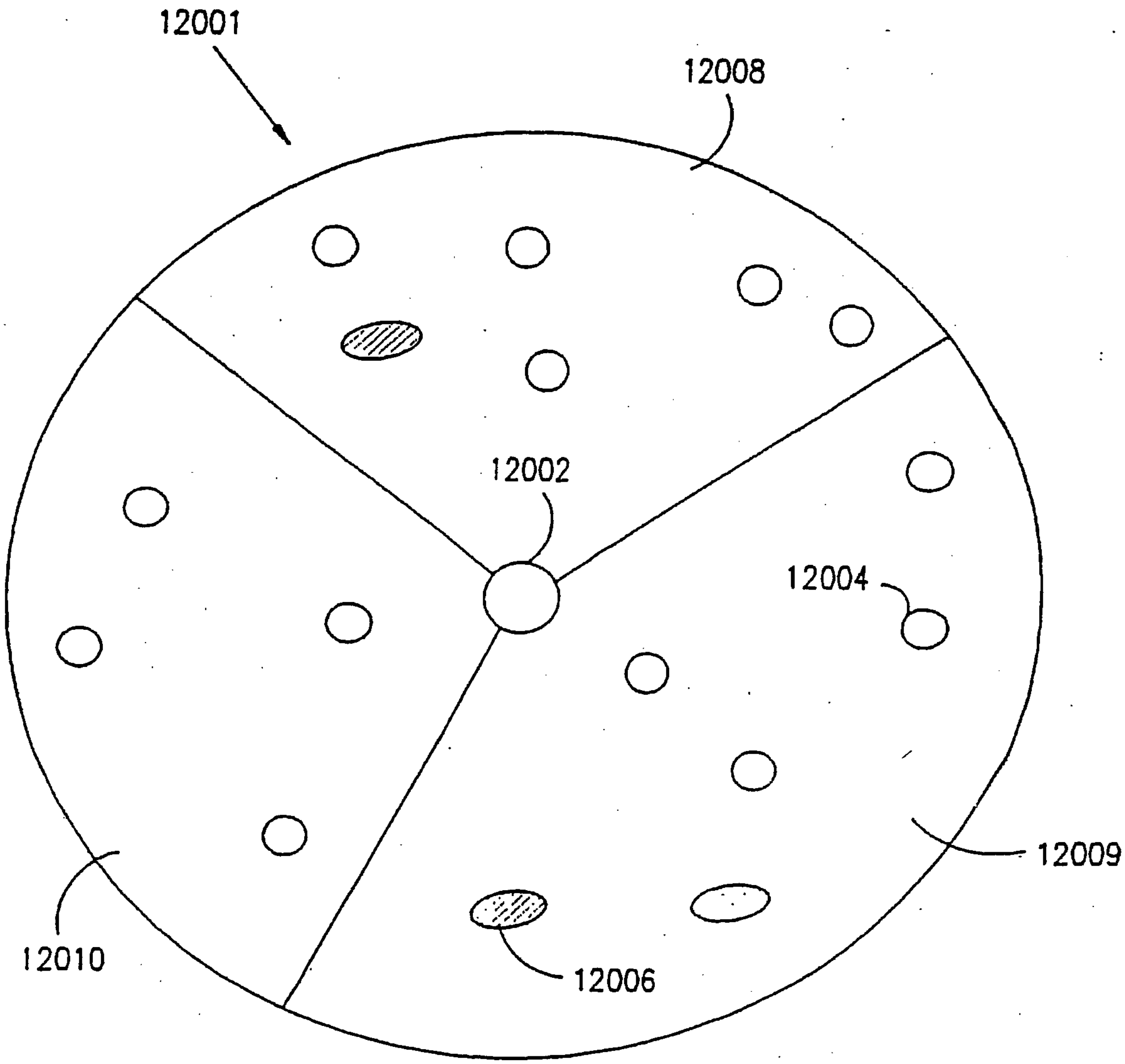
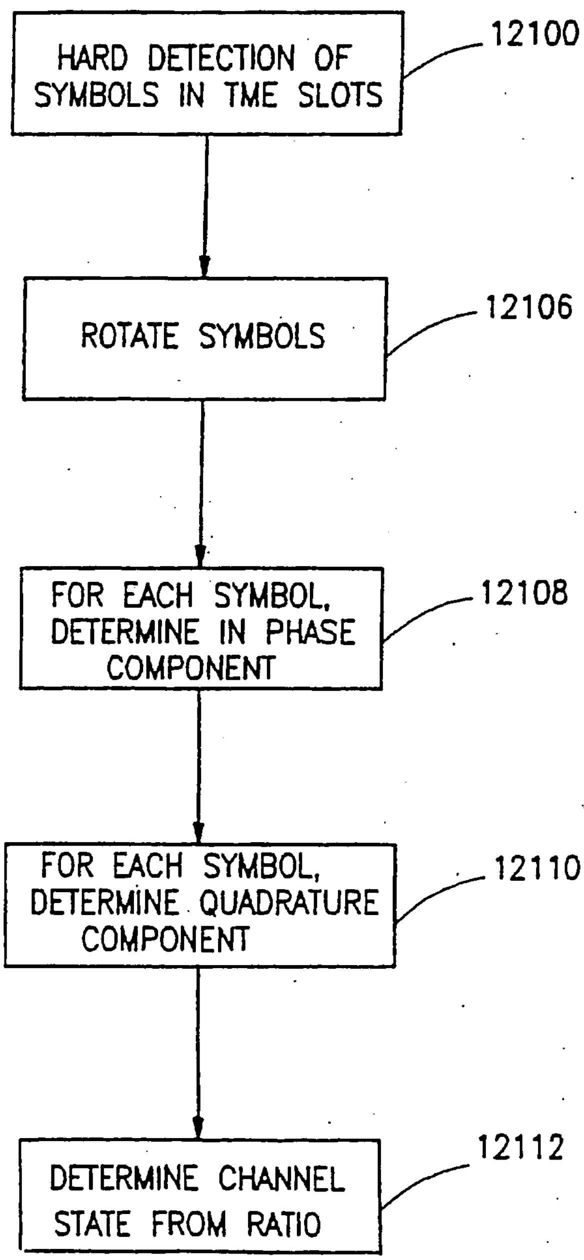


FIG. 98

FIG. 99


HARD DETECTION OF SYMBOLS IN TME SLOTS
12100
ROTATE SYMBOLS
12106
FOR EACH SYMBOL, DETERMINE IN PHASE COMPONENT
12108
FOR EACH SYMBOL, DETERMINE QUADRATURE COMPONENT
12110
DETERMINE CHANNEL STATE FROM RATIO
12112

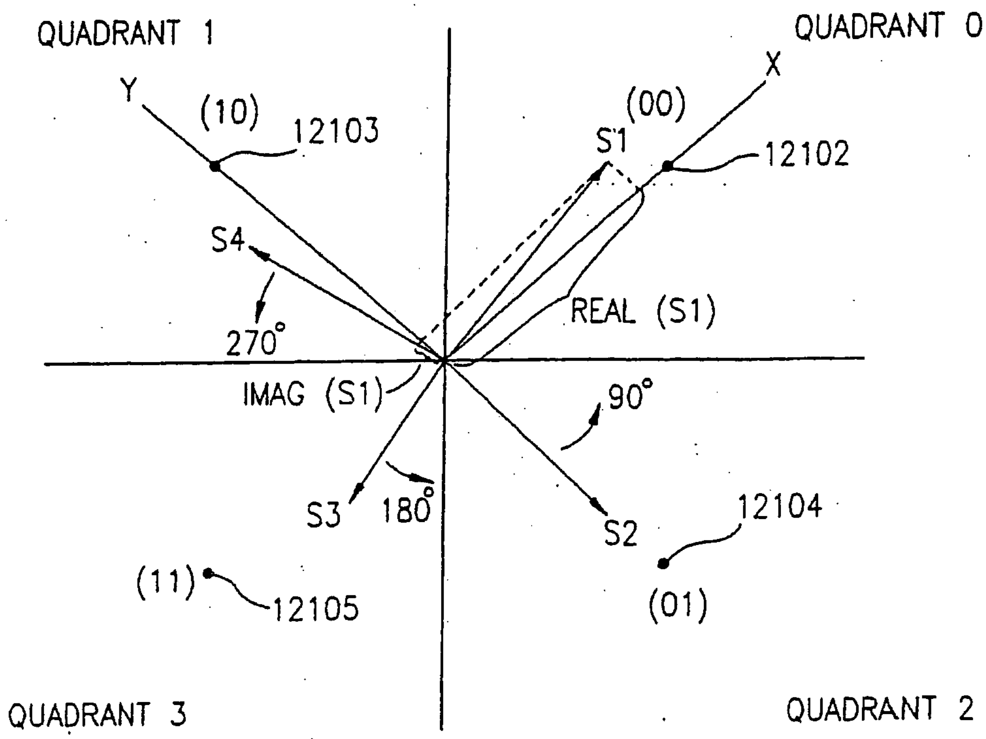
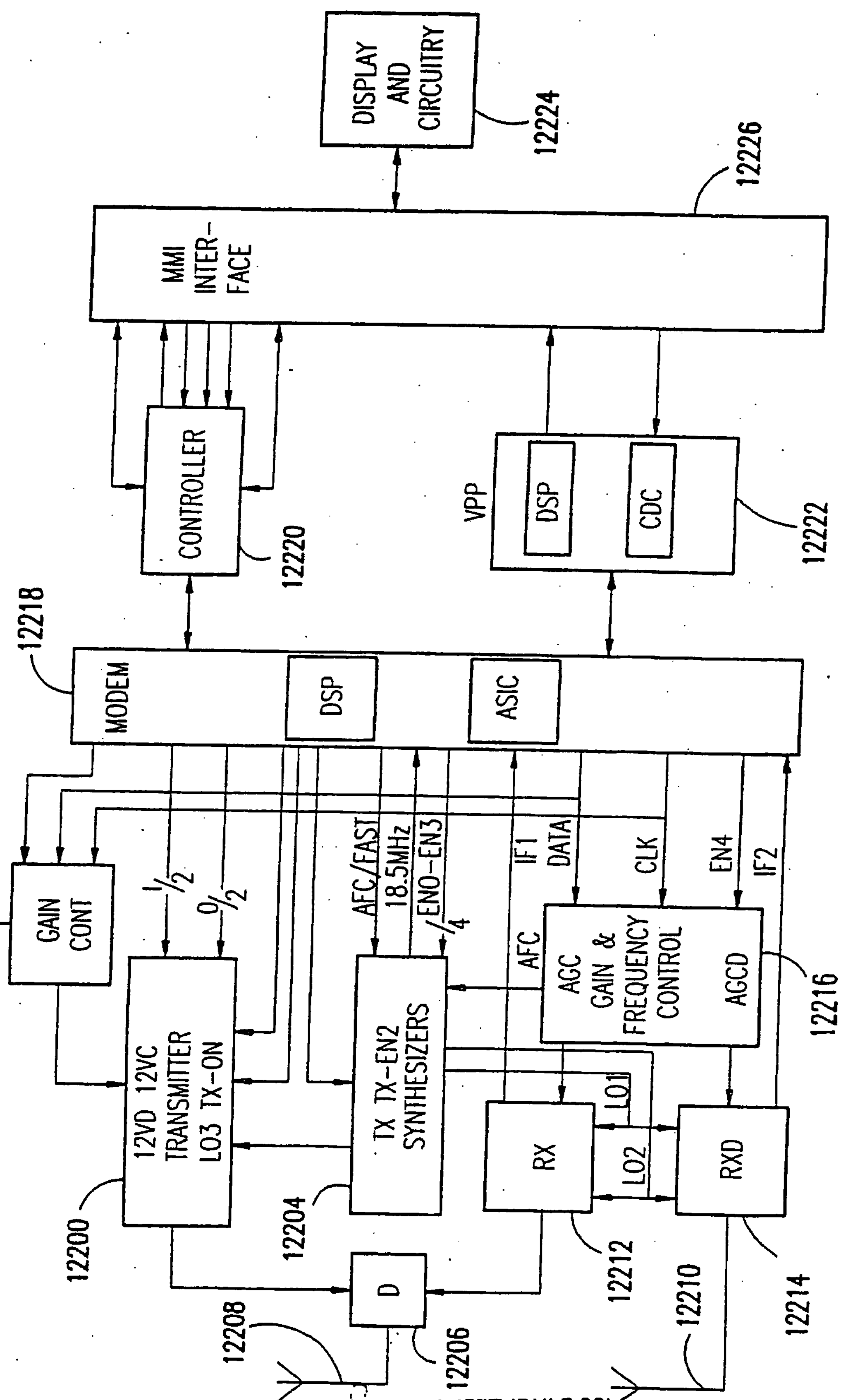

QUADRANT 1
QUADRANT 0
Y
X
(10)
12103
(00)
S1
12102
S4
REAL (S1)
270°
IMAG (S1)
90°
S3
180°
S2
12104
(11)
12105
(01)
QUADRANT 3
QUADRANT 2

FIG. 100

FIG. 101


DISPLAY AND CIRCUITRY
12224
12226
MMI INTER-FACE
CONTROLLER
12220
VPP
DSP
CDC
12222
12218
MODEM
DSP
ASIC
GAIN CONT
12202
1/2
0/2
AFC/FAST
18.5MHz
EN0-EN3
4
IF1
DATA
CLK
EN4
IF2
AFC
AGC
GAIN & FREQUENCY CONTROL
AGCD
12216
12VD 12VC
TRANSMITTER
LO3 TX-ON
12200
TX TX-EN2
SYNTHESIZERS
12204
RX
LO1
LO2
RXD
12212
12214
12210
D
12206
12208

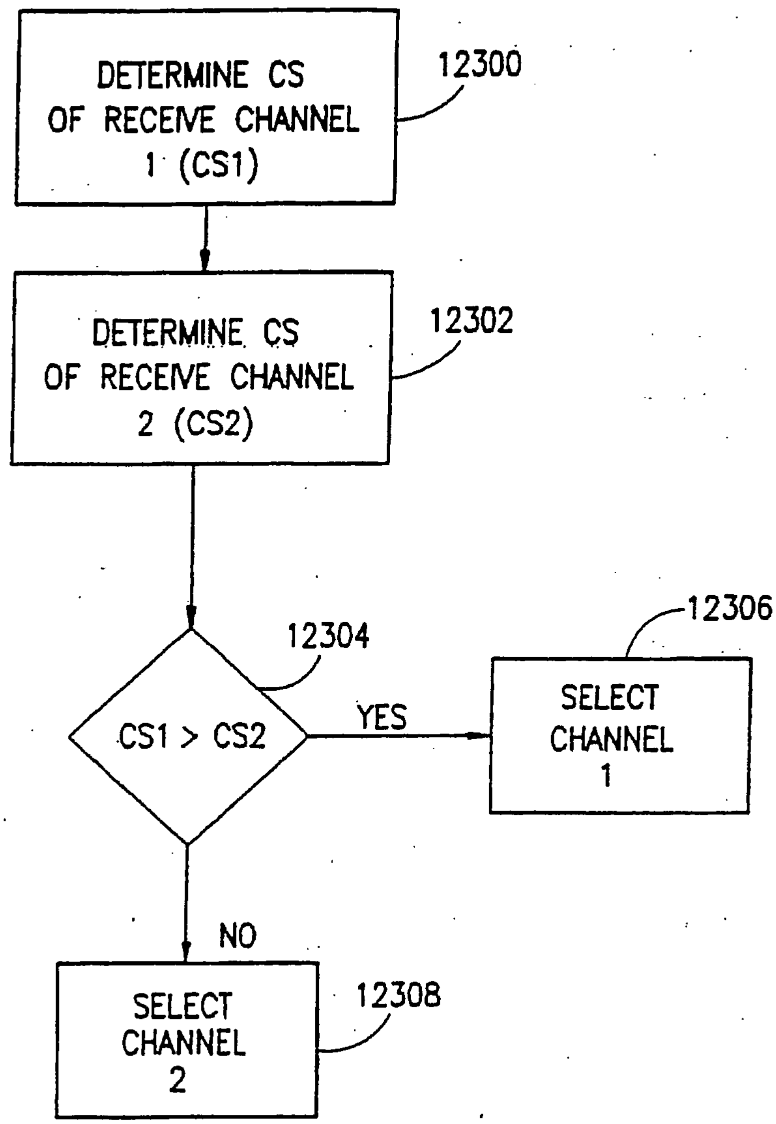

DETERMINE CS
OF RECEIVE CHANNEL
1 (CS1)
12300
DETERMINE CS
OF RECEIVE CHANNEL
2 (CS2)
12302
CS1 > CS2
12304
YES
SELECT
CHANNEL
1
12306
NO
SELECT
CHANNEL
2
12308

FIG. 102

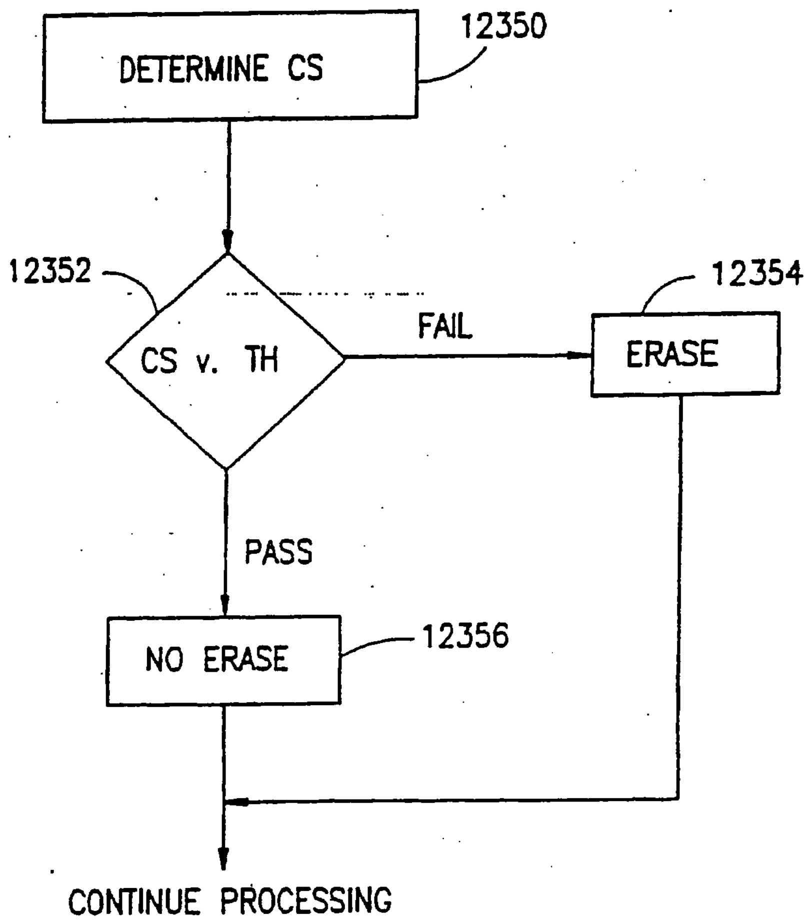

DETERMINE CS
12350
12352
CS v. TH
FAIL
12354
ERASE
PASS
NO ERASE
12356
CONTINUE PROCESSING

FIG. 103

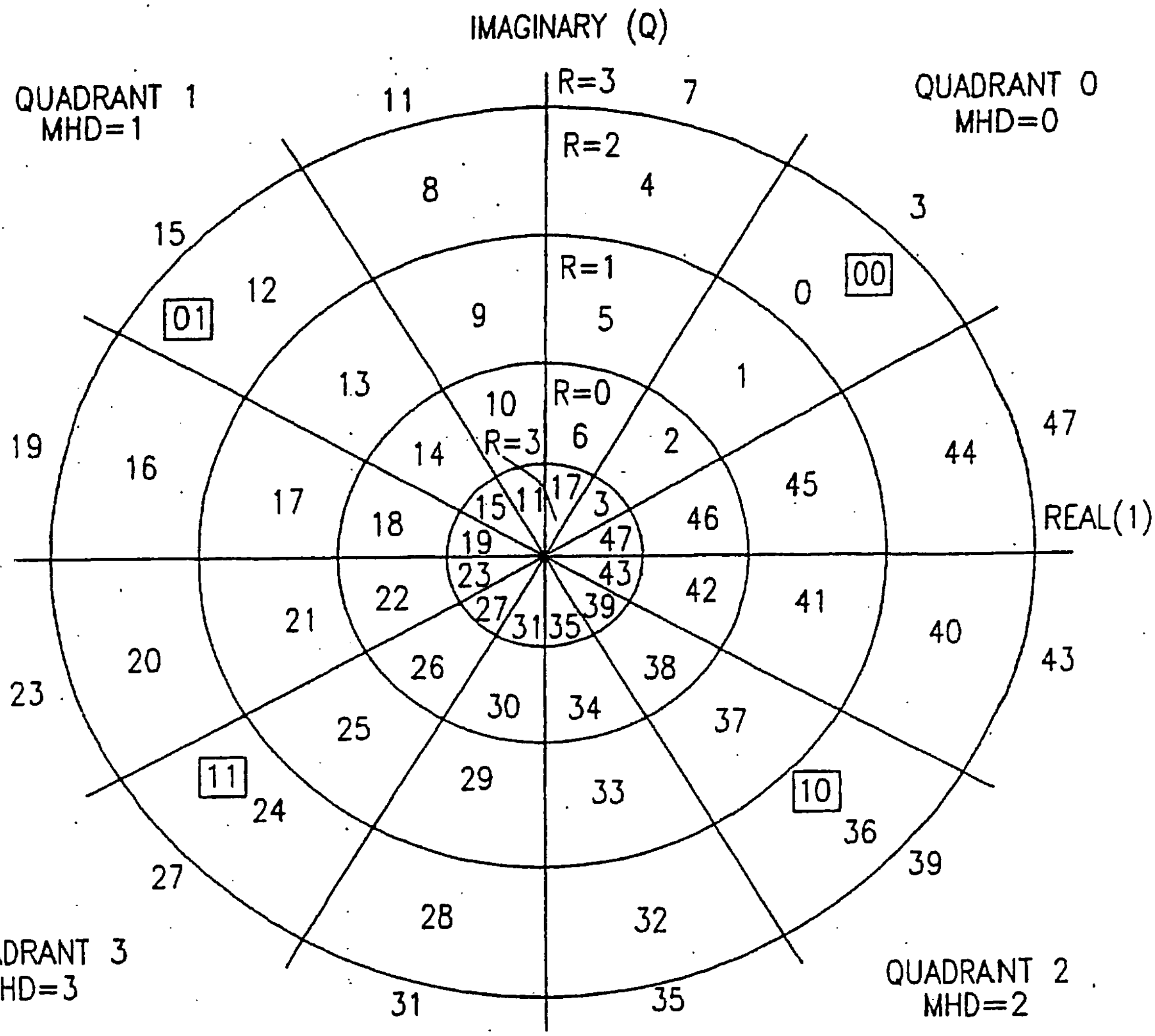

IMAGINARY (Q)
REAL(1)
QUADRANT 0
MHD=0
QUADRANT 1
MHD=1
QUADRANT 2
MHD=2
QUADRANT 3
MHD=3
R=3
R=2
R=1
R=0
R=3
00
01
10
11
0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31 32 33 34 35 36 37 38 39 40 41 42 43 44 45 46 47

FIG. 104

PCT

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114,421	30 June 1995 (30.06.95)	IL
114,422	30 June 1995 (30.06.95)	IL
114,423	30 June 1995 (30.06.95)	IL
114,424	30 June 1995 (30.06.95)	IL
114,425	30 June 1995 (30.06.95)	IL
114,426	30 June 1995 (30.06.95)	IL
114,427	30 June 1995 (30.06.95)	IL
114,428	30 June 1995 (30.06.95)	IL
114,429	30 June 1995 (30.06.95)	IL
114,420	30 June 1995 (30.06.95)	IL
114,419	30 June 1995 (30.06.95)	IL
115,475	1 October 1995 (01.10.95)	IL

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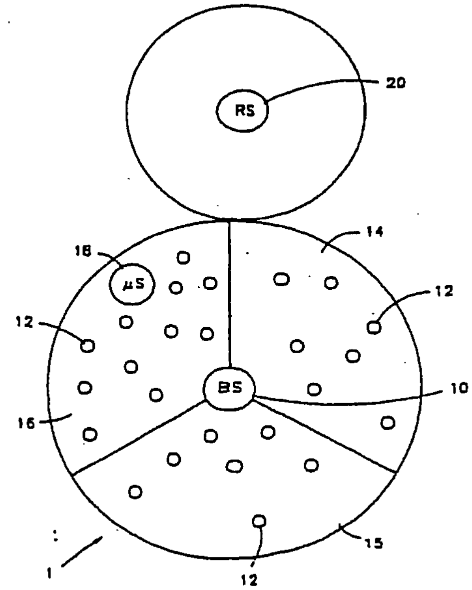
(54) Title: SECTORIZED COMMUNICATION SYSTEM AND METHODS USEFUL THEREFOR

(57) Abstract

Apparatus for providing voice and data in a frequency hopping multiple access communication system (1), preferably including automatic gain control, time alignment, power control, coordinated hits and forced erasures, automatic frequency control, delay loop, fringe area handling, channel feature acquisition and tracking, hand off, and talk around and retransmission.



INTERNATIONAL SEARCH REPORT

International application No.
PCT/US95/13457

A. CLASSIFICATION OF SUBJECT MATTER
IPC(6) :H04B 1/00; H04M 11/00
US CL :Please See Extra Sheet.
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 455/69, 33.1, 33.2, 33.3, 54.1, 56.1, 234.1, 62, 63; 370/20, 24, 25, 95.1, 95.3; 379/59, 60; 375/200, 324, 326

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched
none

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
none

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US, A, 4,613,990 (Halpern) 23 September 1986 (see figures 4-5 and the abstract)	1-4, 7-10, 92-95, 97-104, 106-111
X	US, A, 5,278,992 (Su et al.) 11 January 1994 (see col. 2, lines 5-28)	1-4, 7-10, 92-131
X	US, A, 5,101,501 (Gilhousen et al) 31 March 1992 (see col. 7-8)	5, 79-91
X	US, A, 5,204,977 (Feldt) 20 April 1993 (see figures 1 and 2)	6, 53-72
X	US, A, 5,229,996 (Backstrom et al) 20 July 1993 (see figures 4-5)	132-152

[X] Further documents are listed in the continuation of Box C. [ ] See patent family annex.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be part of particular relevance

"E" earlier document published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
22 MAY 1996	11 JUN 1996

Name and mailing address of the ISA/US	Authorized officer
Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231	Andrew Faile B. Handw for
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Form PCT/ISA/210 (second sheet)(July 1992)★

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US95/13457

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US, A, 5,327,576 (Uddenfeldt et al) 05 July 1994 (see col. 7)	187-215
A	US, A, 5,369,670 (Zagloul et al) 29 November 1994 (see col. 4)	225-252
A	US, A, 5,373,507 (Skold) 13 December 1994 (see figure 4)	225-252
A	US, A, 5,297,169 (Backstrom et al) 22 March 1994 (see abstract)	225-252

Form PCT/ISA/210 (continuation of second sheet)(July 1992)*

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US95/13457

Box I Observations where certain claims were f und unsearchable (Continuation of item 1 of first sheet)

This international report has not been established in respect of certain claims under Article 17(2)(a) for the following reasons:

1. ☐ Claims Nos.:
because they relate to subject matter not required to be searched by this Authority, namely:

2. ☐ Claims Nos.:
because they relate to parts of the international application that do not comply with the prescribed requirements to such an extent that no meaningful international search can be carried out, specifically:

3. ☐ Claims Nos.:
because they are dependent claims and are not drafted in accordance with the second and third sentences of Rule 6.4(a).

Box II Observations where unity of invention is lacking (Continuation of item 2 of first sheet)

This International Searching Authority found multiple inventions in this international application, as follows:

Please See Extra Sheet.

1. ☒ As all required additional search fees were timely paid by the applicant, this international search report covers all searchable claims.

2. ☐ As all searchable claims could be searched without effort justifying an additional fee, this Authority did not invite payment of any additional fee.

3. ☐ As only some of the required additional search fees were timely paid by the applicant, this international search report covers only those claims for which fees were paid, specifically claims Nos.:

4. ☐ No required additional search fees were timely paid by the applicant. Consequently, this international search report is restricted to the invention first mentioned in the claims; it is covered by claims Nos.:

Remark n Protest
☐ The additional search fees were accompanied by the applicant's protest.
☐ No protest accompanied the payment of additional search fees.

A. CLASSIFICATION OF SUBJECT MATTER:
US CL :

455/69, 33.2, 33.3, 234.1, 62, 63; 370/20, 95.3; 379/59, 60; 375/200, 326

BOX II. OBSERVATIONS WHERE UNITY OF INVENTION WAS LACKING
This ISA found multiple inventions as follows:

This application contains the following inventions or groups of inventions which are not so linked as to form a single inventive concept under PCT Rule 13.1. In order for all inventions to be examined, the appropriate additional examination fees must be paid.

Group I, claim(s) 1-4, 7-10 and 92-131, drawn to a power control method, where the transmitter power is regulated via a comparison with the difference between the received power and a reference level.
Group II, claim(s) 5 and 79-91, drawn to handoff for a radio telephone system, where the mobile searches sync information from adjacent sectors and requests a handoff to the largest detected sync signal and the base station establishes a three-way communication link and monitors voice activity therein to determine when to handoff the communication.
Group III, claim(s) 6 and 53-72, drawn to an automatic gain control apparatus in a slotted radio communication system, where the amplitude values are averaged and filtered to adjust the gain of the following time slot.
Group IV, claims 15-16, drawn to a digital communication system with error replacement, where a multi-bit digital signal is divided into a plurality of segments and where only the segments containing errors are replaced by re-transmission.
Group V, claims 17-22, drawn to a frequency control system, where a modem generates a frequency offset and a local oscillator generates a control signal to cancel the frequency offset.GroupVI, claims 23-52, drawn to a self-synchronizing receiver, w^ere an embedded sync code is detected and used to sync with a local timing system.
Group VII, claims 11-14 and 73-78, drawn to an amplification control arrangement maintaining a fixed input to a modem, where a current demodulator input amplitude value is converted into a logarithmic type unit, compared with the modem input amplitude and used to control the amplifier.
Group VIII, claims 132-152, drawn to a time alignment error adjustment, where a time alignment adjustment signal is sent from a first to a second station and used to adjust the receipt of subsequent messages.
Group IX, claims 153-166, drawn to a method for processing received messages in order to reduce repeat transmissions, where transmitted messages include a plurality of sub-messages and incorrectly received sub-messages are repeated and replaced by re-transmitted messages.
Group X, claims 167-186, drawn to a collision avoidance method for a frequency hopping multiple access communication system, where a plurality of frequency channels are provided for transmitting and receiving signals over slots defined with respect to both time and frequency and subscribers are allowed to skip transmissions of at least one slot selected in accordance with a predetermined sequence.
Group XI, claims 187-215, drawn to a hybrid full duplex and half duplex system, where halfduplexlinksareusedbetweensubscriberswhoareproximate to one another and full duplex links are used elsewhere.
Group XII, claims 216-217 and 223, drawn to a method for reducing interference between a plurality of sectors in a communication system, where an area which surrounds a base station is assigned to an individual one of the communication sectors and air resources are allocated within the surrounding area.
Group XIII, claims 218-222 and 224, drawn to a base station with antenna selection, where a plurality of sector antennas are disposed at a given height relative to the ground and an auxiliary antenna is disposed at a height less than the given height and includes a radiation pattern covering an entire azimuthal vicinity of the base station.
Group XIV, claims 225-252, drawn to a method for deriving the state of a channel, where a plurality of signals are received and the in-phase components are determined and summed and the quadrature components are determined and summed and the channel state is determined based on a ratio of the summed in-phase and summed quadrature components.

The inventions listed as Groups I-XIV do not relate to a single inventive concept under PCT Rule 13.1 because, under PCT Rule 13.2, they lack the same or corresponding special technical features for the following reasons: Groups I-XIV are directed toward diverse features of a communication system and operation which are mutually exclusive. The diverse special technical features relevant to eachgroupareset forthinthe precedinggroupings, where none of the groups require the special technical features of any other of the groups.